UNIT II PULSE AND DATA COMMUNICATION

INTRODUCTION:

3.1 Pulse Modulation

The continuous time signal $x(t)$ to be transmitted is sampled at frequency f_s sufficiently above the highest frequen cy present in $x(t)$. The amplitude of the modulating signal $x(t)$ modulates some parameter of the pulse train. These parameters are amplitude, duration (width) and position. Fig. 3.1.1 shows different types of analog pulse modulation techniques with message waveform $x(t)$.

For PAM the modulated pulse parameter is amplitude, for PDM it is width and for PPM it is relative position. These parameters vary in direct proportion to amplitude of $x(t)$ at the sampling instant. As shown in waveforms of Fig. 3.1.1.

$$f_s = \frac{1}{T_s} = \text{Sampling frequency}$$

$$A_0 = \text{Amplitude of the pulse}$$

and $$\tau_0 = \text{Width of the pulse}$$

Since the waveforms are unipolar, they have some dc value. Also the shape of the pulse should be preserved (rising and falling edges, amplitude, duration etc.). Thus the transmission bandwidth needed for these pulse transmission is quite high compared to the message signal bandwidth. Therefore normally single channel PAM, PPM or PDM are seldom used. Always time division multiplexing (TDM) is used to utilize the channel bandwidth.

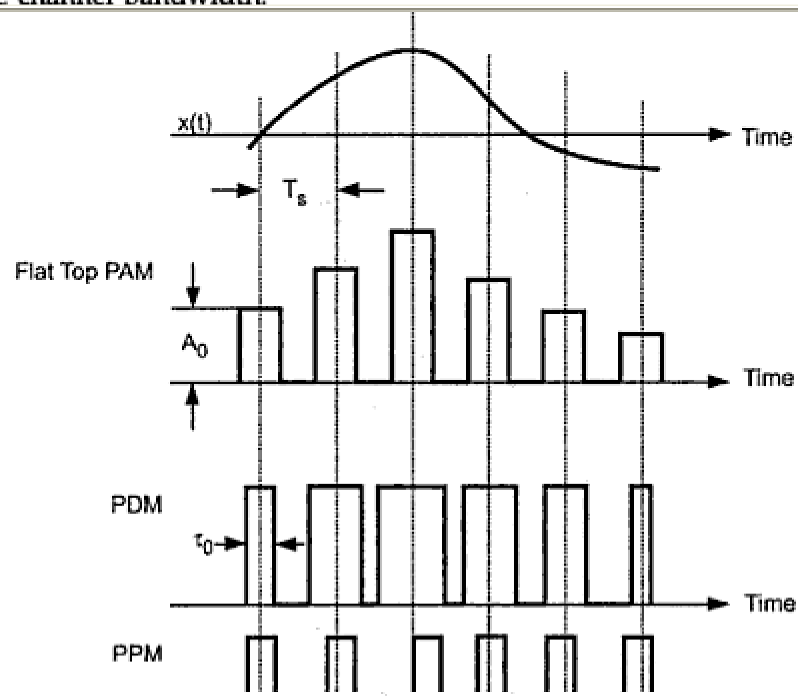

Fig. 3.1.1 Different types of analog pulse modulation techniques

3.2 Pulse Code Modulation (PCM)

3.2.1 PCM Generator

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The pulse code modulator technique samples the input signal $x(t)$ at frequency $f_s \geq 2W$. This sampled 'Variable amplitude' pulse is then digitized by the analog to digital converter. The parallel bits obtained are converted to a serial bit stream. Fig. 3.2.1 shows the PCM generator.

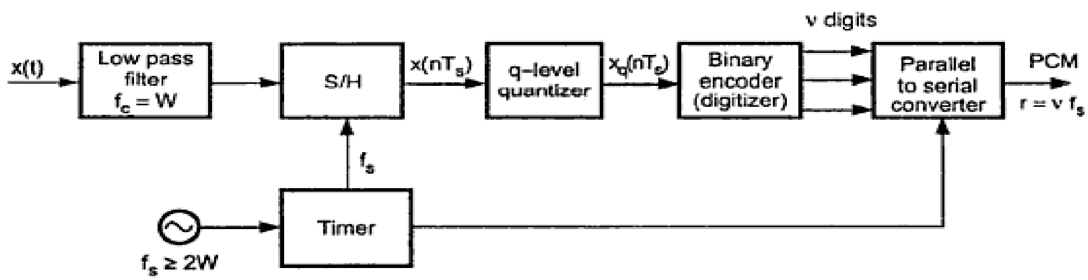

Fig. 3.2.1 PCM generator

In the PCM generator of above figure, the signal $x(t)$ is first passed through the low-pass filter of cutoff frequency 'W' Hz. This low-pass filter blocks all the frequency components above 'W' Hz. Thus $x(t)$ is bandlimited to 'W' Hz. The sample and hold circuit then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficiently above Nyquist rate to avoid aliasing i.e.,

$$f_s \geq 2W$$

In Fig. 3.2.1 output of sample and hold is called $x(nT_s)$. This $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels. It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called *quantization error*. Thus output of quantizer is a digital level called $x_q(nT_s)$.

Quantization error is given as,

$$\varepsilon = x_q(nT_s) - x(nT_s) \qquad \text{... (3.2.1)}$$

3.2.2 Transmission Bandwidth in PCM

Let the quantizer use 'v' number of binary digits to represent each level. Then the number of levels that can be represented by 'v' digits will be,

$$q = 2^v \qquad \text{... (3.2.2)}$$

Here 'q' represents total number of digital levels of q-level quantizer.

For example if $v = 3$ bits, then total number of levels will be,

$$q = 2^3 = 8 \text{ levels}$$

Each sample is converted to 'v' binary bits. i.e. Number of bits per sample = v

We know that, Number of samples per second = f_s

∴ Number of bits per second is given by,

$$\begin{aligned}\text{(Number of bits per second)} &= \text{(Number of bits per samples)} \\ &\quad \times \text{(Number of samples per second)} \\ &= v \text{ bits per sample} \times f_s \text{ samples per second}\end{aligned}$$

3.2.3 PCM Receiver

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Fig. 3.2.2 (a) shows the block diagram of PCM receiver and Fig. 3.2.2 (b) shows the reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulses and removes the noise. This signal is then converted to parallel digital words for each sample.

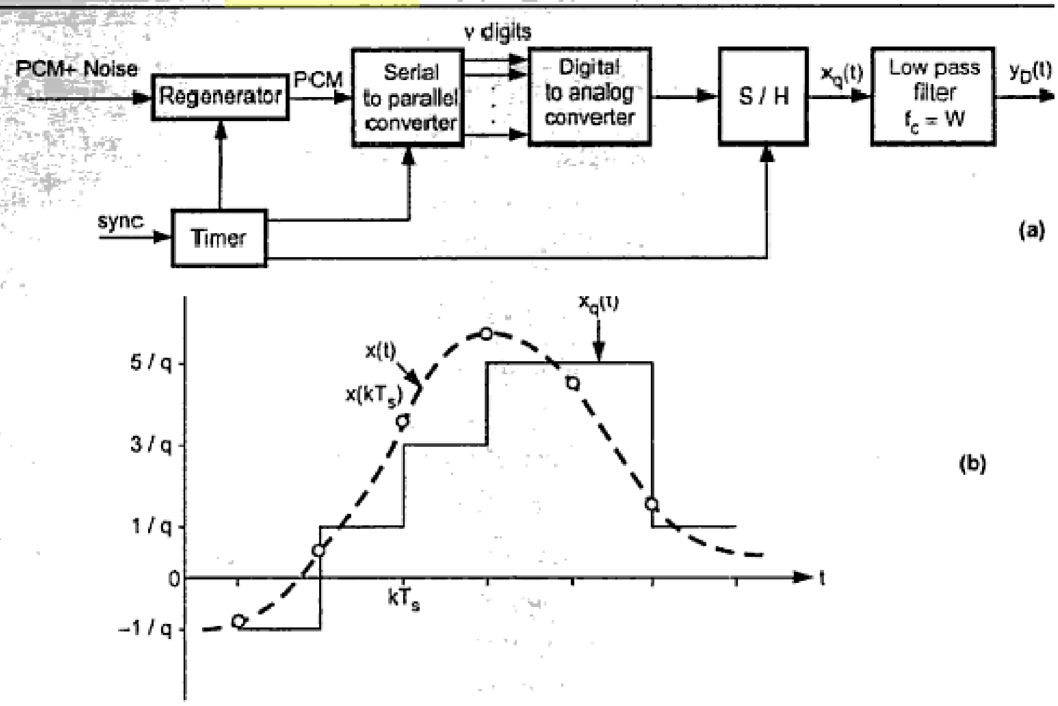

Fig. 3.2.2 (a) PCM receiver
(b) Reconstructed waveform

The digital word is converted to its analog value $x_q(t)$ along with sample and hold. This signal, at the output of S/H is passed through lowpass reconstruction filter to get $y_D(t)$. As shown in reconstructed signal of Fig. 3.2.2 (b), it is impossible to reconstruct exact original signal $x(t)$ because of permanent quantization error introduced during quantization at the transmitter. This quantization error can be reduced by increasing the binary levels. This is equivalent to increasing binary digits (bits) per sample. But increasing bits 'v' increases the signaling rate as well as transmission bandwidth as we have seen in equation (3.2.4) and equation (3.2.7).

1.8.6.3 Companding in PCM

Normally we don't know how the signal level will vary in advance. Therefore the nonuniform quantization (variable step size 'δ') becomes difficult to implement. Therefore the signal is amplified at low signal levels and attenuated at high signal levels. After this process, uniform quantization is used. This is equivalent to more step size at low signal levels and small step size at high signal levels. At the receiver a reverse process is done. That is signal is attenuated at low signal levels and amplified at high signal levels to get original signal. Thus the compression of signal at transmitter and expansion at receiver is called combinely as *companding*. Fig. 1.8.9 shows compression and expansion curves.

As can be seen from Fig. 1.8.9, at the receiver, the signal is expanded exactly opposite to compression curve at transmitter to get original signal. A dotted line in the Fig. 1.8.9 shows uniform quantization. The compression and expansion is obtained by passing the signal through the amplifier having nonlinear transfer characteristic as

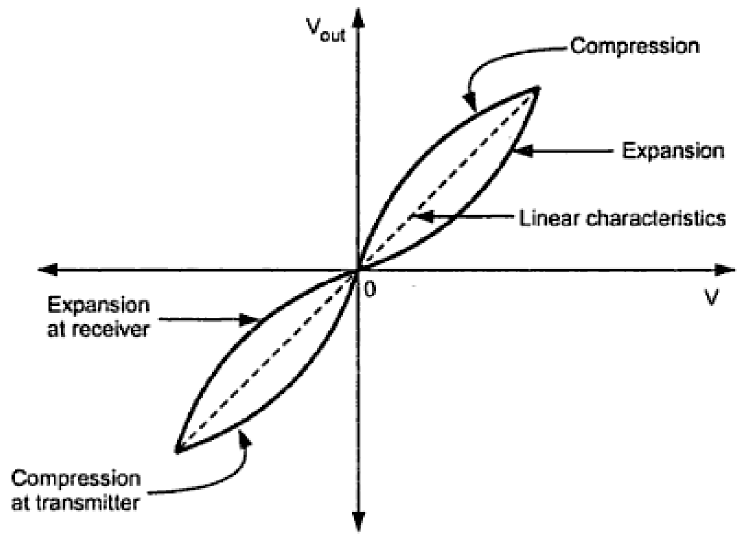

Fig. 1.8.9 Companding curves for PCM

1.8.6.4 μ - Law Companding for Speech Signals

Normally for speech and music signals a μ - law compression is used. This compression is defined by the following equation,

$$Z(x) = (\text{Sgn } x)\frac{ln(1+\mu|x|)}{ln(1+\mu)} \quad |x| \leq 1 \qquad \text{... (1.8.52)}$$

Fig. 1.8.10 shows the variation of signal to noise ratio with respect to signal level without companding and with companding.

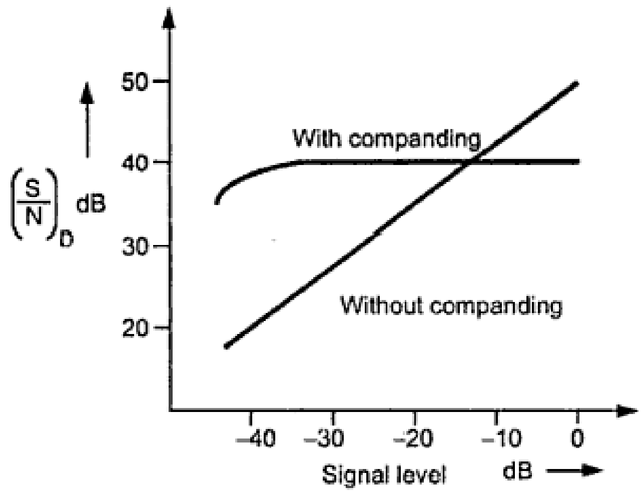

Fig. 1.8.10 PCM performance with μ - law companding

1.8.6.5 A-Law for Companding

The A law provides piecewise compressor characteristic. It has linear segment for low level inputs and logarithmic segment for high level inputs. It is defined as,

$$Z(x) = \begin{cases} \dfrac{A|x|}{1+\ln A} & \text{for } 0 \le |x| \le \dfrac{1}{A} \\ \dfrac{1+\ln(A|x|)}{1+\ln A} & \text{for } \dfrac{1}{A} \le |x| \le 1 \end{cases} \qquad \ldots (1.8.53)$$

When A = 1, we get uniform quantization. The practical value for A is 87.56. Both A-law and μ-law companding is used for PCM telephone systems.

Delta Modulation (DM) :

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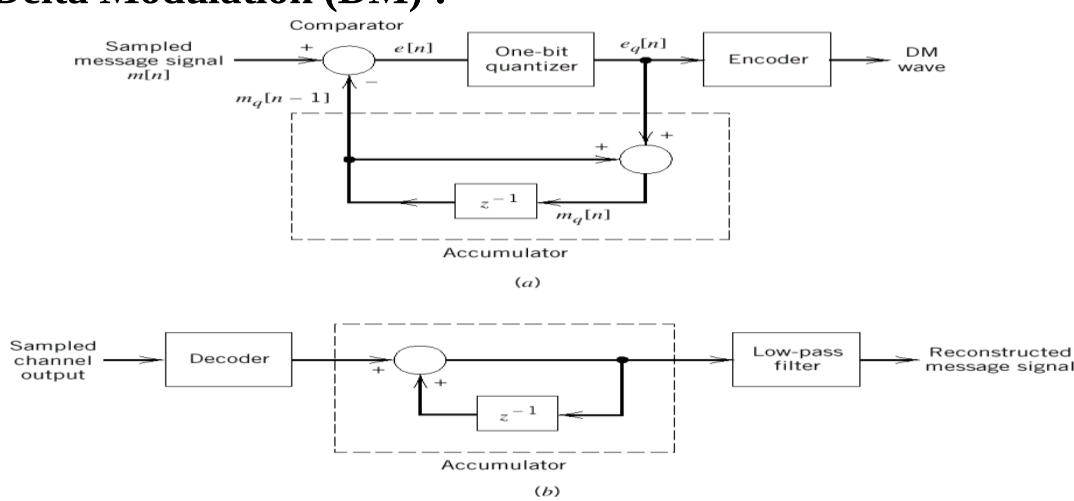
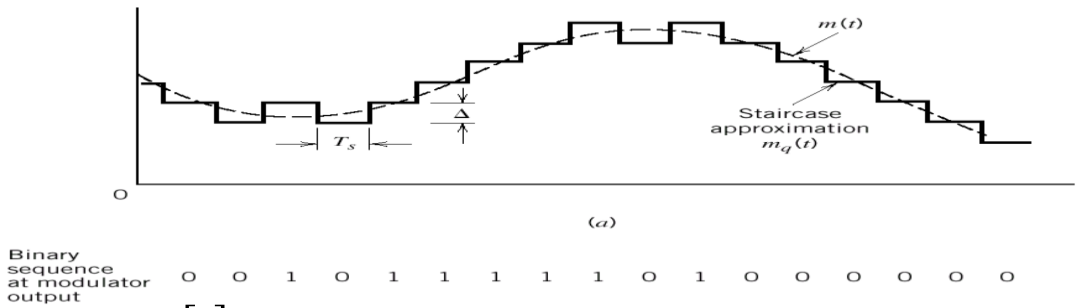




Let $m[n]=m(nT_s)\ ,\ n=0,\pm1,\pm2,\ldots$

where T_s is the sampling period and $m(nT_s)$ is a sample of $m(t)$.

The error signal is

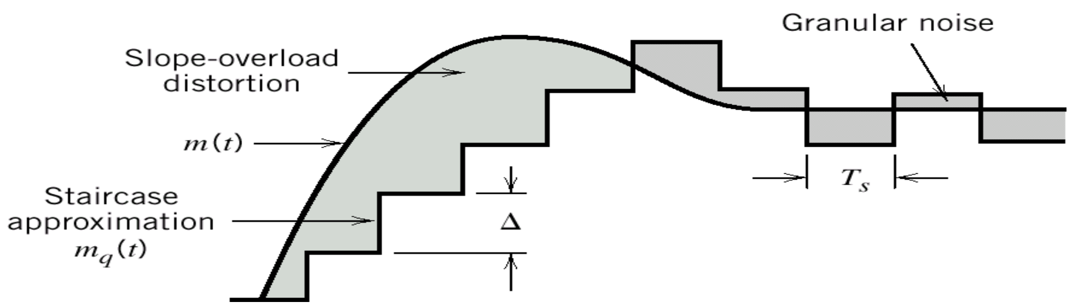
$$e[n]=m[n]-m_q[n-1] \qquad (3.52)$$

$$e_q[n]=\Delta \operatorname{sgn}(e[n]) \qquad (3.53)$$

$$m_q[n]=m_q[n-1]+e_q[n] \qquad (3.54)$$

where $m_q[n]$ is the quantizer output, $e_q[n]$ is the quantized version of $e[n]$, and Δ is the step size

The modulator consists of a comparator, a quantizer, and an accumulator. The output of the accumulator is



Differential Pulse-Code Modulation (DPCM):

Usually **PCM** has the sampling rate higher than the **Nyquist rate** .The encode signal contains redundant information. **DPCM** can efficiently remove this redundancy**.**

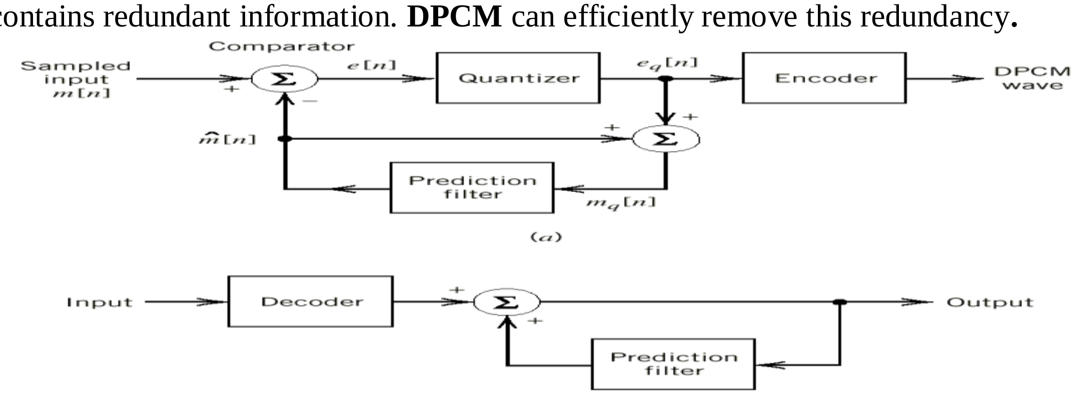


Figure 3.28 DPCM system. (*a*) Transmitter. (*b*) Receiver.

$$e[n] = m[n] - \hat{m}[n] \qquad (3.74)$$

$\hat{m}[n]$ is a prediction value.

The quantizer output is

$$e_q[n] = e[n] + q[n] \qquad (3.75)$$

where $q[n]$ is quantization error.

The prediction filter input is

$$m_q[n] = \underbrace{\hat{m}[n] + e[n]}_{m[n]} + q[n] \qquad (3.77)$$

$$\Longrightarrow m_q[n] = m[n] + q[n] \qquad (3.78)$$

Processing Gain:

The $(\mathrm{SNR})_o$ of the DPCM system is

$$(\mathrm{SNR})_o = \frac{\sigma_M^2}{\sigma_Q^2} \qquad (3.79)$$

where σ_M^2 and σ_Q^2 are variances of $m[n]$ $(E[m[n]] = 0)$ and $q[n]$

$$(\mathrm{SNR})_o = \left(\frac{\sigma_M^2}{\sigma_E^2}\right)\left(\frac{\sigma_E^2}{\sigma_Q^2}\right)$$

$$= G_p (\mathrm{SNR})_Q \qquad (3.80)$$

where σ_E^2 is the variance of the predictions error

and the signal to - quantization noise ratio is

-

$$(\mathrm{SNR})_Q = \frac{\sigma_E^2}{\sigma_Q^2} \qquad (3.81)$$

Processing Gain, $G_p = \dfrac{\sigma_M^2}{\sigma_E^2}$ (3.82)

Design a prediction filter to maximize G_p (minimize σ_E^2)

TABLE OF CONTENTS

CHAPTER 1

INTRODUCTION TO DATA COMMUNICATIONS

- **COMPUTER NETWORK**

Interconnected collection of autonomous computers that are able to exchange information.

- No master/slave relationship between computers in the network.

- **DATA COMMUNICATIONS**

Transmission of signals in a reliable and efficient matter.

- **COMMUNICATION MODEL (SYSTEM)**

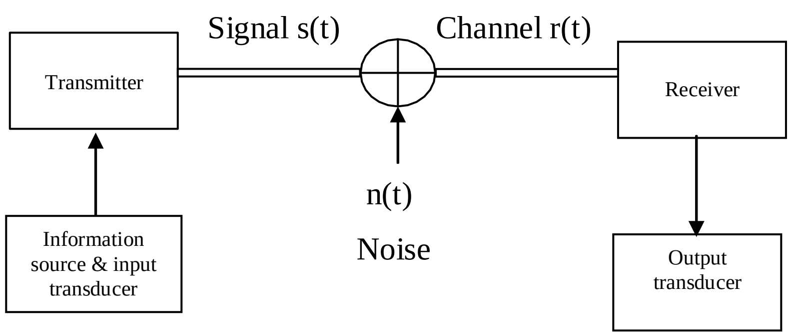
The purpose of a communications system is to exchange data between two entities.

- **Source:** entity that generates data; eg. a person who speaks into the phone, or a computer sending data to the modem.
- **Transmitter:** a device to transform/encode the signal generated by the source.
  - the transformed signal is actually sent over the transmission system.
  eg. a modem transforms digital data to analog signal that can be handled by the telephone network.
- **Transmission System (Channel):** medium that allows the transfer of a signal from one point to another.
  eg. a telephone network for a computer/modem.
- **Receiver:** a device to decode the received signal for handling by destination device.
  eg. A modem converts the received analog data back to digital for the use by the computer.
- **Destination:** entity that finally uses the data.
  eg. Computer on other end of a receiving modem.

Data Communications

Data communications is the transfer of information that is in digital form, before it enters the communication system.

- Basic Elements of a Communication System



Basic Elements of a Communication System

- **Information:** generated by the source may be in the form of voice, a picture or a plain text. An essential feature of any source that generates information is that its output is described in probabilistic terms; that is, the output is not deterministic.

  A transducer is usually required to convert the output of a source in an electrical signal that is suitable for transmission.

- **Transmitter:** a transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. In general, a transmitter performs the matching of the message signal to the channel by a process called modulation.

  The choice of the type of modulation is based on several factors, such as:

  - the amount of bandwidth allocated,

  - the type of noise and interference that the signal encounters in transmission over the channel,

  - and the electronic devices that are available for signal amplification prior to transmission.

- **Channel:** the communication channel is the physical medium that connects the transmitter to the receiver. The physical channel may be a pair of wires that carry the electrical signals, or an optical fibre that carries the information on a modulated light beam or free space at which the information-bearing signal are electromagnetic waves.

- **Receiver:** the function of a receiver is to recover the message signal contained in the received signal. The main operations performed by a receiver are demodulation, filtering and decoding.

Analog and Digital Data Transmission

- Data are entries that convey information.
- Signals are electrical encoding (representation) of data.
- Signalling is the act of propagation of signals through a suitable medium.

The terms analog and digital correspond to continuous and discrete, respectively. These two terms are frequently used in data communications.

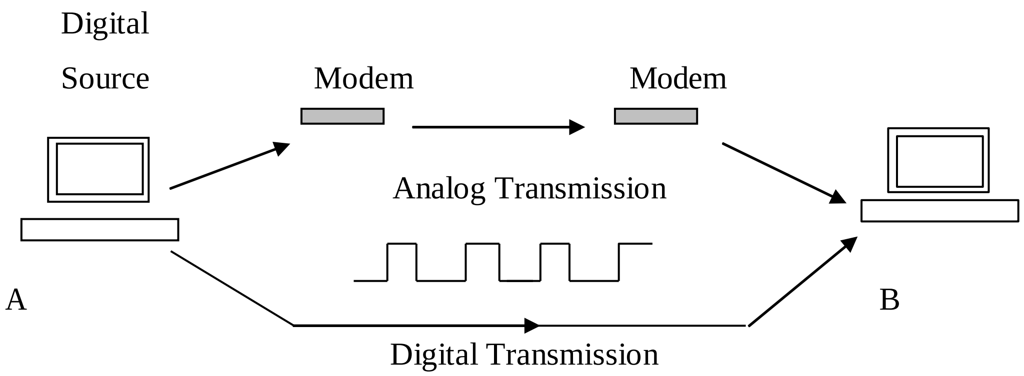
Analog data takes on continuous values on some interval. The most familiar example of analog data is audio signal. Frequency components of speech may be found between

20 Hz and 20 kHz. The basic speech energy is concentrated between 300-3400 Hz. The frequencies up to 4000 Hz add very little to the intelligibility of human ear.
Another common example of analog data is video. The outputs of many sensors, such as temperature and pressure sensors, are also examples of analog data.

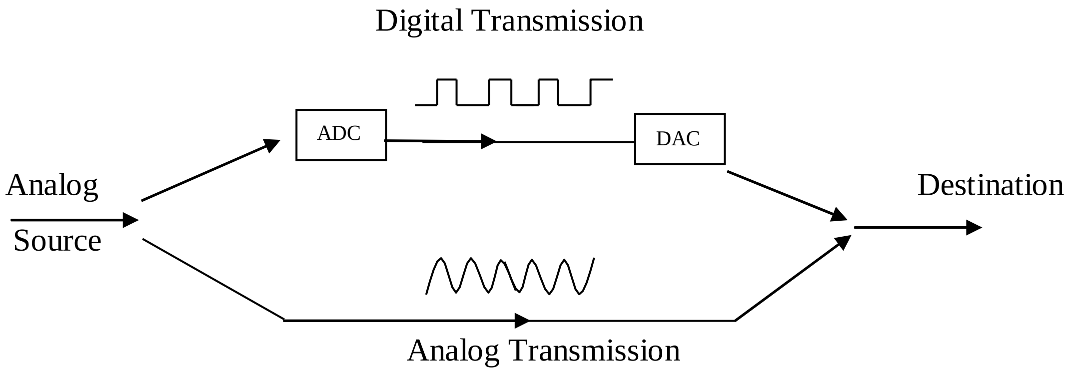
Digital data takes on discrete values; eg. a computer's output.

- Analog transmission is a means of transmitting analog signals regardless of their content. The data may be analog or digital.
- Digital transmission is the transfer of information through a medium in digital form. A digital signal can be transmitted only for a limited distance.
- Data communications is the transfer of information that is in digital form, before it enters the communication system.

• Two methods of sending data from computer A to computer B. both cases are examples of data communications, because the original data is digital in nature.



• Two ways of transmitting analog information. In either cases it is not data communications, because the original information is not digital.



ADC: Analog-Digital-Converter

DAC: Digital-Analog-Converter

Digital Communication System

Up to this point, we have described an electrical communication system in rather broad terms based on the implicit assumption that the message signal is a continuous time-varying waveform. We refer to such continuous-time signal waveforms as analog signals and to the corresponding information sources that produce such signals as analog sources. Analog signals can be transmitted directly via carrier modulation over the communication channel and demodulated accordingly at the receiver. We call such a communication system an analog communication system.

Alternatively, an analog source output may be converted into a digital form and the message can be transmitted via digital modulation and demodulated as a digital signal at the receiver. There are some potential advantages to transmitting an analog signal by means of digital modulation. The most important reason is that signal fidelity is better controlled through digital transmission than analog transmission. In particular, digital transmission allows us to regenerate the digital signal in long-distance transmission, thus eliminating effects of noise at each regeneration point. In contrast, the noise added in analog transmission is amplified analog with the signal when amplifiers are used periodically to boost the signal level in long-distance transmission. Another reason for choosing digital transmission over analog is that the analog message signal may be highly redundant. With digital processing, redundancy may be removed prior to modulation, thus conserving channel bandwidth. Yet a third reason may be that digital communication systems are often cheaper to implement.

In some applications, the information to be transmitted is inherently digital, e.g., in the form of English text, computer data, etc. In such cases, the information source that generates the data is called a discrete (digital) source.

In a digital communication system, the functional operations performed at the transmitter and receiver must be expanded to include message signal discrimination at the transmitter and message signal synthesis or interpolation at the receiver. Additional functions include redundancy removal, and channel coding and decoding.

Figure 1.2 illustrates the functional diagram and the basic elements of a digital communication system. The source output may be either an analog signal, such as audio or video signal, or a digital signal, such as the output of a Teletype machine, which is discrete in time and has a finite number of output characters. In a digital communication system, the messages produced by the source are usually converted into a sequence of binary digits.

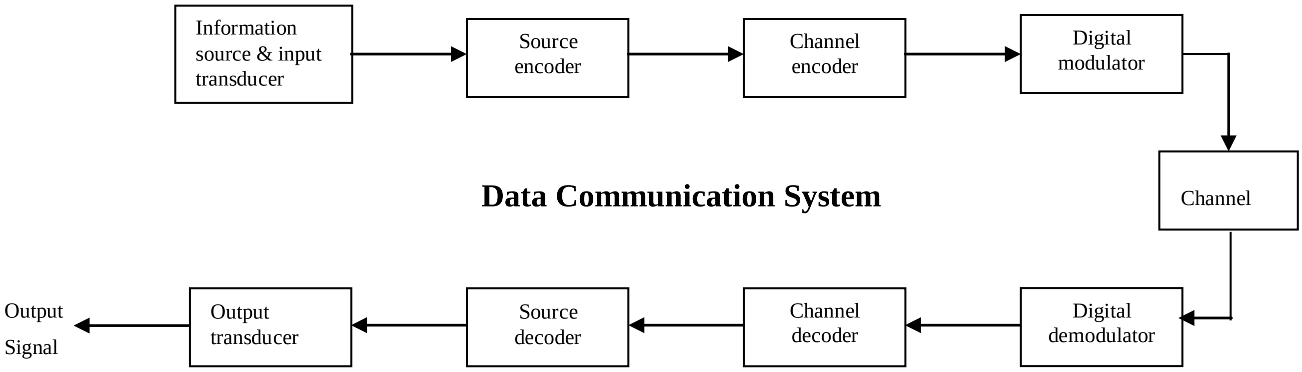


Figure 1.2. Basic elements of a digital communication system

Ideally, we would like to represent the source output (message) by as few binary digits as possible. In other words, we seek an efficient representation of the source output that results in little or no redundancy. The process of efficiently converting the output of either an analog or a digital source into a sequence of binary digits is called source encoder or data compression.

The sequence of binary digits from the source encoder, which we call the information sequence, is passed to the channel encoder. The purpose of the channel encoder is to introduce in a controlled manner some redundancy in the binary information sequence, which can be used at the receiver to overcome the effects of noise and interference encountered in the transmission of the signal through the channel. Thus, the added redundancy serves to increase the reliability of the received data and improves the fidelity of the received signal. In effect, redundancy in the information sequence aids the receiver in decoding the desired information sequence. For example, a (trivial) form of encoding of the binary information sequence is simply to repeat each binary digit m times, where m is some positive integer. More sophisticated (nontrivial) encoding involves taking k information bits at a time and mapping each k-bit sequence into a unique n-bit sequence, called a code word. The amount of redundancy introduced by encoding the data in this manner is measured by the ratio n/k. The reciprocal of this ratio, namely, k/n is called the rate of the code or, simply, the code rate.

The binary sequence at the output of the channel encoder is passed to the digital modulator, which serves as the interface to the communications channel. Since nearly all of the communication channels encountered in practice are capable of transmitting electrical signals (waveforms), the primary purpose of the digital modulator is to map the binary information sequence into signal waveforms. To elaborate on this point, let us suppose that the coded information sequence is to be transmitted one bit at a time at some uniform rate R bits/s. The digital modulator may simply map the binary digit 0 into a waveform $s_0(t)$ and the binary digit 1 into a waveform $s_1(t)$. In this manner, each bit from the channel encoder is transmitted

separately. We call this binary modulation. Alternatively, the modulator may transmit b coded information bits at a time by using M =2b distinct waveform $s_i(t)$, I = 0, 1, …, m-1, one waveform for each of the 2b possible b-bits sequences. We call this M-ary modulation (M >2). Note that a new b-bit sequence enters the modulator every b/R seconds. Hence, when the channel bit rate R is fixed, the amount of time available to transmit one of the M waveforms corresponding to a b-bit sequence is b times the period in a system that uses binary modulation.

At the receiving end of a digital communications system, the digital demodulator processes the channel-corrupted transmitted waveform and reduces each waveform to a single number that represents an estimate of the transmitted data symbol. For example, when binary modulation is used, the demodulator may process the received waveform and decide on whether the transmitted bit is a 0 or 1. In such a case, we say the demodulator has made a binary decision. As one alternative, the demodulator may make a ternary decision; that is, it decides that the transmitted bit is either a 0 or 1 or it makes no decision at all, depending on the apparent quality of the received signal. When no decision is made on a particular bit, we say that the demodulator has inserted an erasure in the demodulated data. Using the redundancy in the transmitted data, the decoder attempts to fill in the positions where erasures occurred. Viewing the decision process performed by the demodulator as a form of quantization, we observe that binary and ternary decisions are special cases of a demodulator that quantizes to Q levels, where $Q \geq 2$ In general, if the digital communications system employs M-ary modulation, where m = 0,1, … , M represent the M possible transmitted symbols, each corresponding to $k = \log_2 M$ bits, the demodulator may make A Q-ary decision, where $Q \geq M$ .In the extreme case where no quantization is performed, $Q = \infty$.

When there is no redundancy in the transmitted information, the demodulator must decide which of the M waveforms was transmitted in any given time interval. Consequently, Q = M, and since there is no redundancy in the transmitted information, no discrete channel decoder is used following the demodulator. On the other hand, when there is redundancy introduced by a discrete channel encoder at the transmitter, the Q-ary output from the demodulator occurring every k/R seconds is fed to the decoder, which attempts to reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained in the received data. A measure of how well the demodulator and encoder perform is the frequency with which errors occur in the decoded sequence. More precisely, the average probability of a bit-error at the output of the decoder is a measure of the performance of the demodulator-decoder combination. In general, the probability of error is a function of the code characteristics, the types of waveforms used to transmit the information

over the channel, the transmitter power, the characteristics of the channel (i.e., the amount of noise), the nature of the interference, etc., and the method of demodulation and decoding. These items and their effect on performance will be discussed in detail in subsequent chapters.

As a final step, when an analog output is desired, the source decoder accepts the output sequence from the channel decoder, and from knowledge of the source encoding method used, attempts to reconstruct the original signal from the source. Due to channel decoding errors and possible distortion introduced by the source encoder and, perhaps, the source decoder, the signal at the output of the source decoder is an approximation to the original source output. The difference or some function of the difference between the original signal and the reconstructed signal is a measure of the distortion introduced by the digital communications system.

Early Work in Digital Communications

Although Morse is responsible for the development of the first electrical digital communication system (telegraphy), the beginnings of what we now regard as modem digital communications stem from the work of Nyquist (1924), who investigated the problem of determining the maximum signalling rate that can be used over a telegraph channel of a given bandwidth without intersymbol interference. He formulated a model of a telegraph system in which a transmitted signal has the general form

$$S(t) = \sum_n a_n g(t - nT)$$

Where $g(t)$ represents a basic pulse shape and $\{a_n\}$ is the binary data sequence of $\{\pm 1\}$ transmitted at a rate of $1/T$ bits per second. Nyquist set out to determine the optimum pulse shape that was bandlimited to W Hz and maximised the bit rate $1/T$ under the constraint that the pulse caused no intersymbol interference at the sampling times k/T, $k = 0, \pm 1, \pm 2,$. His studies led him to conclude that the maximum pulse rate$1/T$ is $2W$ pulses per second. This rate is now called the Nyquist rate. Moreover, this pulse rate can be achieved by using the pulses $g(t) = (\sin 2\pi\ Wt)/2\pi\ Wt$. This pulse shape allows the recovery of the data without intersymbol interference at the sampling instants Nyquist's result is equivalent to a version of the sampling theorem for bandlimited signals, which was later stated precisely by Shannon (1948). The sampling theorem states that a signal of bandwidth W can be reconstructed from samples taken at the Nyquist rate of $2W$ samples per second using the interpolation formula

$$S(t) = \sum_n s\left(\frac{n}{2W}\right) \frac{\sin 2\pi W(t - n/2W)}{2\pi W(t - n/2W)}$$

In light of Nyquist's work Hartley (1928) considered the issue of the amount of data that can be transmitted reliably over a bandlimited channel when multiple amplitude levels are used. Due to the presence of noise and other interference, Hartley postulated that the receiver could reliably estimate the received signal amplitude to some accuracy, say A_δ. This investigation led Hartley to conclude that there is maximum data rate that can be communicated reliably over a bandlimited channel when the maximum signal amplitude is limited to A_{max} (fixed power constraint) and the amplitude resolution is A_δ.

Another significant advance in the development of communications was the work of Wiener (1942) who considered the problem of estimating a desired signal waveform $s(t)$ in the presence of additive noise $n(t)$, based on observation of the received signal $r(t) = s(t) + n(t)$. This problem arises in signal demodulation. Wiener determined the linear filter whose output is the best mean-square approximation to the desired signal $s(t)$. The resulting filter is called the optimum linear (Wiener) filter. Hartley's and Nyquist results on the maximum transmission rate of digital information were precursors to the work of Shannon (1948 a, b) who established the mathematical foundations for information theory and derived the fundamental limits for digital communication systems. In his pioneering work, Shannon formulated the basic problem of reliable transmission of information in statistical terms, using probabilistic models for information sources and communication channels. Based on such a statistical formulation, he adopted a logarithmic measure for the information content of a source. He also demonstrated that the effect of a transmitter power constraint, a bandwidth constraint, and additive noise can be associated with the channel and incorporated into a single parameter, called the channel capacity For example, in the case of an additive white (spectrally flat) Gaussian noise interference, an ideal bandlimited channel of bandwidth W has a capacity C given by

$$C = W\,log_2\left(1 + \frac{P}{WN_0}\right) \text{ bits/s}$$

where P is the average transmitted power and N_0 is the power spectral density of the additive noise. The significance of the channel capacity is as follows: If the information rate R from the source is less than C ($R < C$), then it is theoretically possible to achieve reliable (error-free) transmission through the channel by appropriate coding. On the other hand, if $R > C$, reliable transmission is not possible regardless of the amount of signal processing performed at the transmitter and receiver. Thus, Shannon established basic limits on communication of information and gave birth to a new field that is now called information theory.

Initially the fundamental work of Shannon had a relatively small impact on the design and development of new digital communications systems. In part, this was due to the small

demand for digital information transmission during the 1950's. Another reason was the relatively large complexity and, hence, the high cost of digital hardware required to achieve the high efficiency and high reliability predicted by Shannon's theory.

Another important contribution to the field of digital communications is the work of Kotelnikov (1947), which provided a coherent analysis of the various digital communication systems based on a geometrical approach. Kotelnikov approach was later expanded by Wozencraft and Jacobs (1965).

The increase in the demand for data transmission during the last three decades, coupled with the development of more sophisticated integrated circuits, has led to the development of very efficient and more reliable digital communications systems. In the course of these developments, Shannon's original results and the generalization of his results on maximum transmission limits over a channel and on bounds on the performance achieved have served as benchmarks for any given communications system design. The theoretical limits derived by Shannon and other researchers that contributed to the development of information theory serve as an ultimate goal in the continuing efforts to design and develop more efficient digital communications systems.

Following Shannon's publications name the classic work of Hamming (1950) on error detecting and error-correcting codes to combat the detrimental effects of channel noise. Hamming's work stimulated many researchers in the years that followed, and a variety of new and powerful codes were discovered, many of which are used today in the implementation of modem communication systems.

CHAPTER II
DATA TRANSMISSION & SIGNALS

Data Transmission

Concepts and Terminology

- **Transmission Terminology**

  Transmission from transmitter to receiver goes over some transmission medium using electromagnetic waves.

  - Guided Media: waves are guided along a physical path; twisted pair, optical fibre, coaxial cable.

  - Unguided Media: waves are not guided; air waves radio waves.

  - Direct Link: signal goes from transmitter to receiver without intermediate devices, other than amplifiers and repeaters.

  - Point-to Point Link: guided media with direct link between two devices.

  - Multipoint Guided Configuration: more than two devices can share the same medium.

- **Frequency, Spectrum, & Bandwidth**

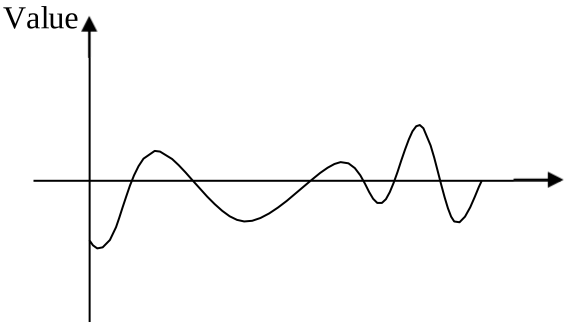
- Signal is generated by a transmitter and transmitted over a medium.
- Signal is a function of time or frequency.

A signal is any function that carries information. Based on the range of variation of independent variables, signals can be divided into two classes: continuous-time (or analog) signals and discrete-time (or digital) signals. A signal is a function of time, but can also be expressed as a function of frequency; that is, the signal consists of components of different frequencies.

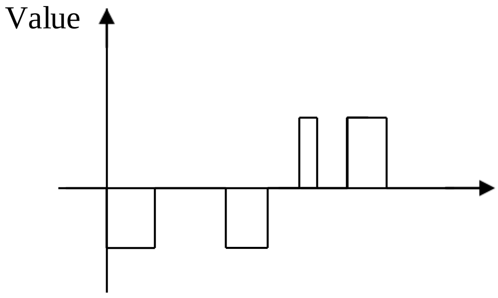
- **Analog and Digital Signals**

Information can be analog or digital. Analog information is continuous. Digital information is discrete.

Signals can be analog or digital. Analog signals can have any value in a range; while digital signals can have only a limited number of values.



Analog Signal



Digital Signal

- **Time –Domain Concepts**

- **Continuous Signal:** Signal intensity varies in a smooth fashion over time; no breaks or discontinuities in the signal.
- **Discrete Signals:** Signal intensity can take one of two pre-specified values for any amount of time.

A continuous time signal is defined by a continuous independent variable. A signal $s(t)$ is continuous if

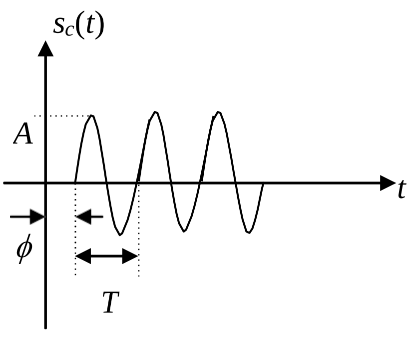
$$\lim_{t \to a} s_c(t) = s(t) \qquad \text{for all a.}$$

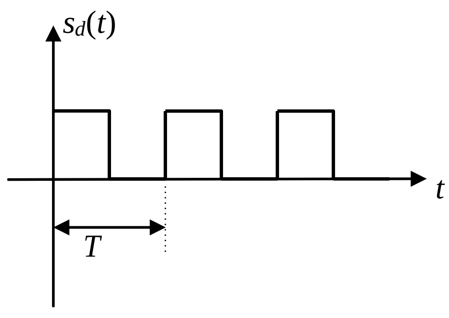
- **Periodic Signal**

A signal $s(t)$ is periodic if

$$s(t + T) = s(t)$$

where T is the period of the signal.





Three important characteristics of a periodic signal are: amplitude, frequency, and phase. Amplitude (A) is the instantaneous value of a signal at any time, and is measured in volts. Frequency (f) is the inverse of the period (T); ($f=1/T$), or the number of period repetition in one second, and is measured in cycles per second or Hertz (Hz). Phase (ϕ) is a measure of

the relative position in time within a single period of a signal. Thus, we can express a sinusoid signal as

$$s(t) = A\sin(2\pi ft + \phi)$$

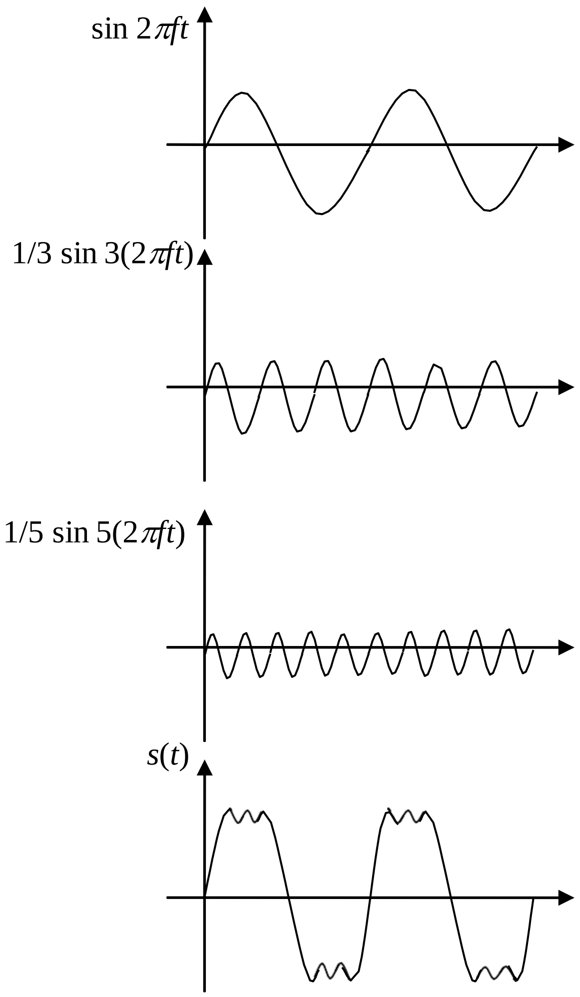
where A is the amplitude, f is the frequency, and ϕ is the phase.

- **Frequency-Domain Concepts**

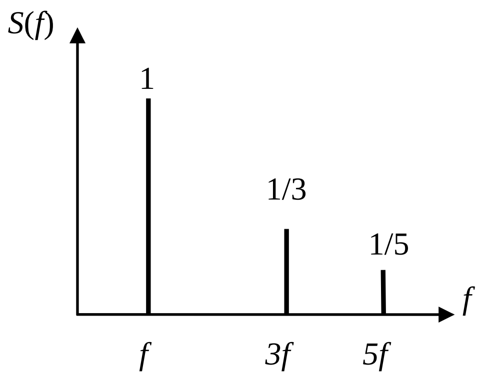
Any signal can also be viewed as a function of frequency, for example, the signal

$$s(t) = \sin 2\pi ft + 1/3\sin 3(2\pi ft) + 1/5\sin 5(2\pi ft)$$

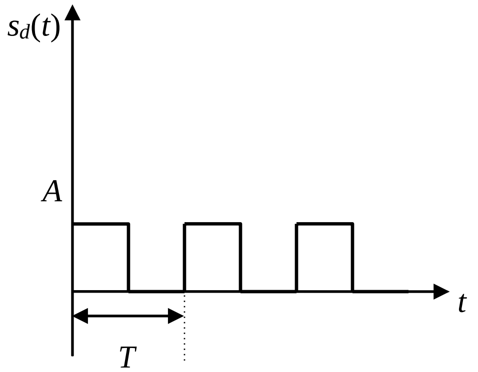
consists of three components as shown in the figure below:



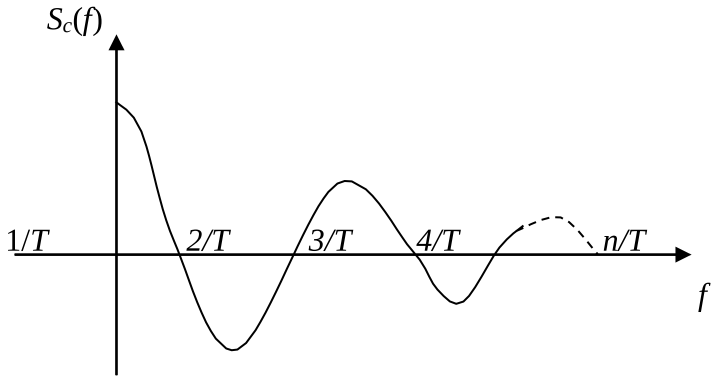
The frequency components of a signal can be determined using Fourier analysis. The following figure shows the spectrum $S(f)$ of the signal $s(t)$. The spectrum of a signal is the range of frequencies that it contains. For this signal the spectrum extends from f to $5f$. the spectrum in this case is discrete.



Many signals, such as the one in the following figure, have continuous spectrum $S_c(f)$ and



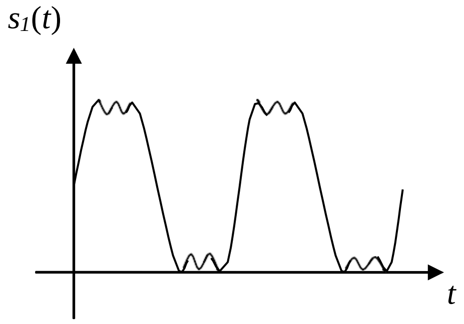
and an infinite bandwidth as shown below:



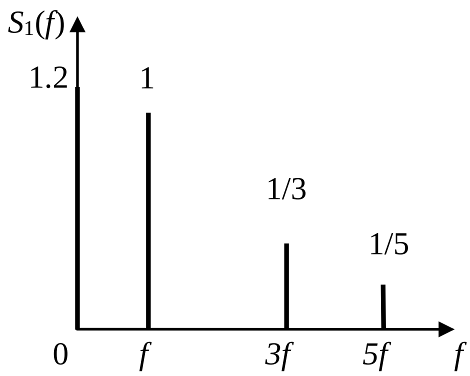
However, most of the energy in the signal is contained in a relatively narrow band of frequencies. This band is referred to as the effective bandwidth, or just bandwidth.

If a signal includes a component of zero frequency, that component is called d_c component or constant component.

The signal $s_1(t)$ in the following figure is obtained by adding a d_c component on $s(t)$:



With a d_c component, it has a frequency term at $f = 0$ and a non-zero average amplitude.



Spectrum of $s_1(t)$

$$s_1(t) = 1.2 + \sin(2\pi ft) + 1/3\sin 3(2\pi ft) + 1/5\sin 5(2\pi ft)$$

- **Fundamental Frequency**

Base frequency such that the frequency of all components can be expressed as its integer multiples; the period of the aggregate signal is the same as the period of the fundamental frequency:

- Each signal can be decomposed into a set of sinusoid signals by making use of Fourier's analysis.
- The time-domain function $s(t)$ specifies a signal in terms of its amplitude at each instant of time.
- The frequency-domain function $S(f)$ specifies the signal in terms of peak amplitude of constituent frequencies.

Spectrum

Range of frequencies contained in a signal.

Absolute Bandwidth

Width of the spectrum.

Effective Bandwidth

Narrow band of frequencies containing most of the energy of the signal.

DC Component

Component of zero frequency; changes the average amplitude of the signal to non-zero.

Relationship between Data Rate and Bandwidth

- Any transmitter/receiver system can accommodate only a limited range of frequencies.
  * The range for FM radio transmission is 88-108 MHz
- This limits the data rate that can be carried over the transmission medium.
- Consider a square wave. Suppose that we let the positive pulse to be binary 1 and the negative pulse to be binary 0. Then, the waveform represents the binary stream 1010… and duration (period) of each pulse is $1/2f$. Thus, the data rate is equal to $2f$ bits per second (bps) or the data rate is equal to twice the fundamental frequency of the digital signal. It can be shown that the frequency-domain representation of this waveform is:

$$s(t) = \sum_{k=1}^{\infty} \frac{1}{k} \sin(2\pi kft)$$

- This waveform has infinite number of frequency components and infinite bandwidth.
- Peak amplitude of the k^{th} frequency component is $1/k$, so most of the energy is concentrated in the first few frequencies.

Ex

Consider a digital transmission system capable of transmitting signals with a bandwidth of 4 MHz.

Case 1

Approximate the square wave with a waveform of the first three sinusoidal components

$$\sin(2\pi ft) + 1/3\sin(2\pi(3f)t) + 1/5\sin(2\pi(5f)t)$$

If $f = 10^6$ cycles per second, or 1 MHz, the bandwidth of the signal

$$s(t) = \left[\sin(2\pi\times10^6 t) + 1/3\sin(2\pi\times3\times10^6 t) + 1/5\sin(2\pi\times5\times10^6 t)\right]$$

is $5\text{x}10^6 - 10^6 = 4$ MHz

For f = 1 MHz, the period of the fundamental frequency is T = $1/10^6$ = 1μs. If the waveform is a bit string of 1's and 0's, then one bit occurs every 0.5 μs for a data rate of 2×10^6 bps or 2 Mbps.

Case 2

Assume a bandwidth of 8 MHz and f = 2 MHz; this gives us the signal bandwidth as

$$(5 \times 2 \times 10^6)-(2 \times 10^6) = 8 \text{ MHz}$$

But T = $1/f$ = 0.5 μs, so that the time needed for one bit is 0.25 μs, giving a data rate of 4 Mbps. Other things being equal, doubling the bandwidth doubles the potential data rate.

Case 3

Let us represent the signal by the first two components of the sinusoid as

$$\left[\sin(2\pi ft) + 1/3\sin(2\pi(3f)t)\right]$$

Assume that f = 2 MHz and T = $1/f$ = 0.5 μs so that the time needed for one bit is 0.25 μs, giving a data rate of 4 Mbps.

Bandwidth of the signal is

$$(3 \times 2 \times 10^6)-(2 \times 10^6) = 4 \text{ MHz}$$

A given bandwidth can support various data rates depending on the ability of the receiver to differentiate between 0 and 1 in the presence of noise and other impairments.

Ex

If a periodic signal is decomposed into five waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is the bandwidth of the signal?

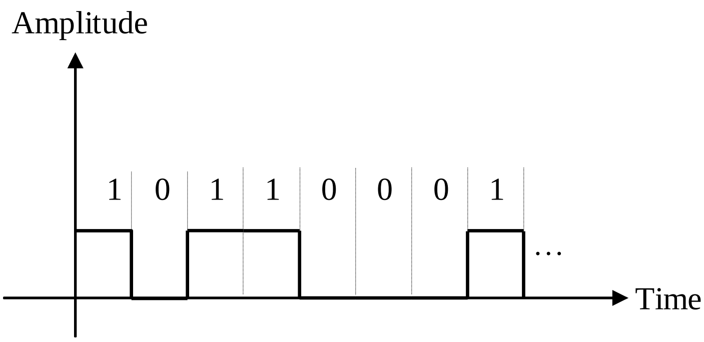
Let f_h be the highest frequency, f_l be the lowest frequency, and B be the bandwidth, then

$$B = f_h - f_l$$

$$= 900-100 = 800 \text{ Hz}$$

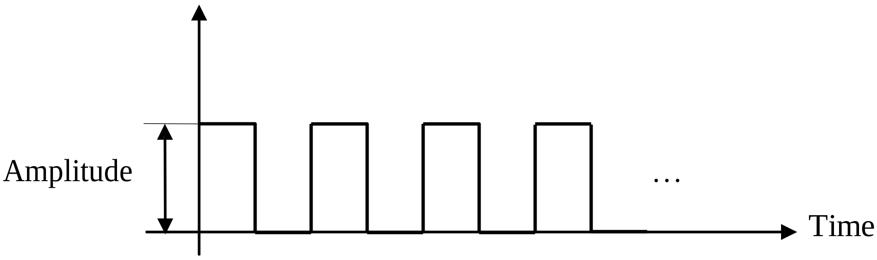
Digital Signals

Data can be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as a zero voltage.

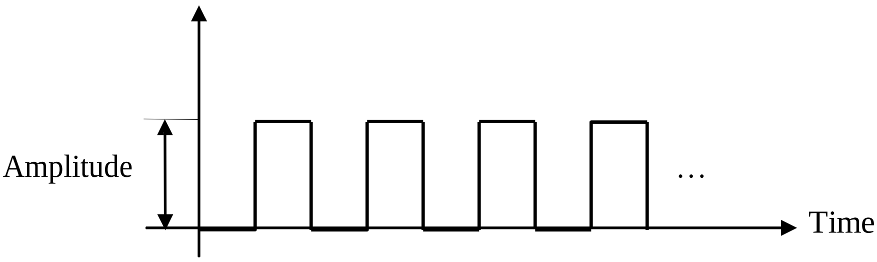


Amplitude, Period and Phase

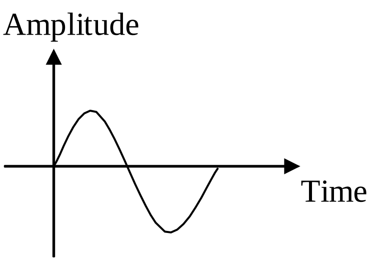
The three characteristics of periodic signals can be redefined for a periodic digital signal



No phase shift

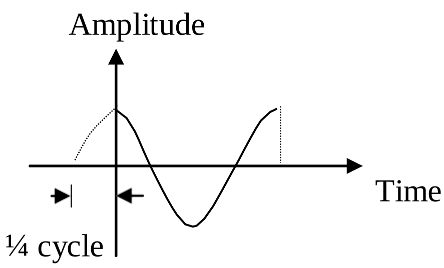


180° phase shift

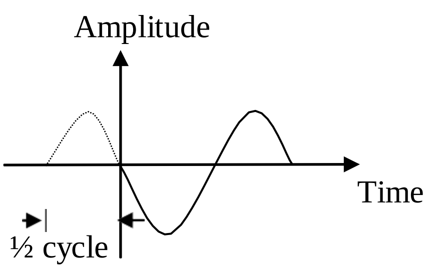


0° phase shift

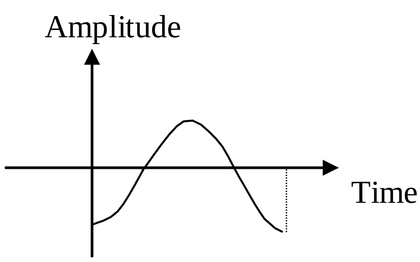
(no phase shift)



90° phase shift



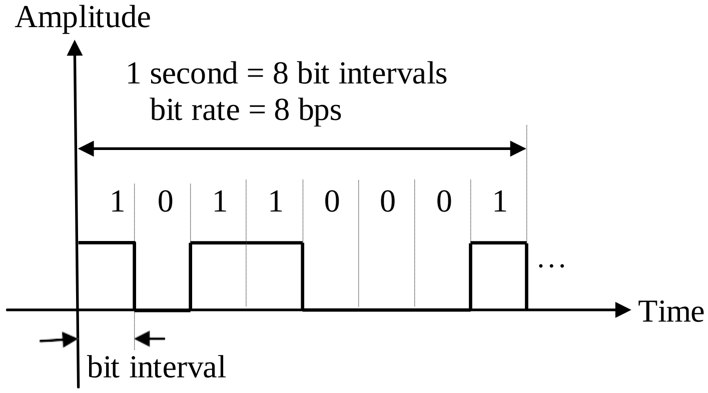
180° phase shift



270° phase shift

Bit Interval and Bit Rate

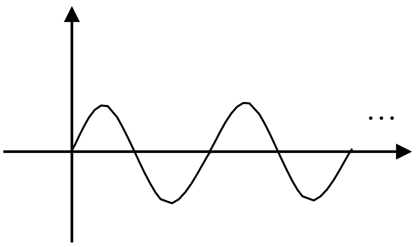
Most digital signals are aperiodic and thus terms like period or frequency are not appropriate. Two new terms, bit interval (instead of period) and bit rate (instead of frequency) are used to describe a digital signal. The bit interval is the time required to send one single bit. The bit rate is the number of bit intervals per second. This means that the bit rate is the number of bits sent in one second, usually expressed in bits per second (bps).



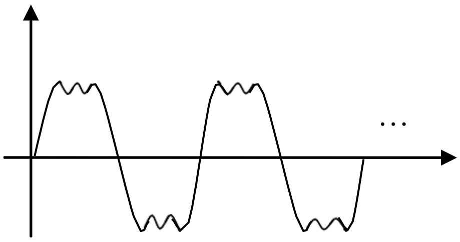
Decomposition of a Digital Signal

A digital signal can be decomposed into an infinite number of simple sine waves called harmonics, each with different amplitude, frequency and phase. This means that when a digital signal is sent along a transmission medium, an infinite number of simple signals is being sent.

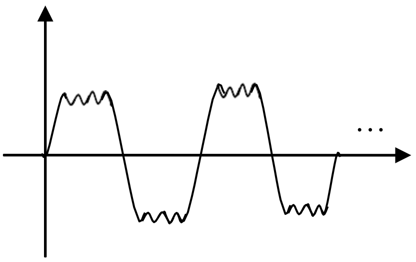
Harmonics of a Digital Signal



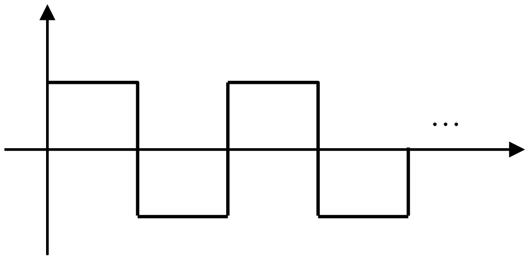
a) First harmonic only



b) First, third, and fifth harmonics

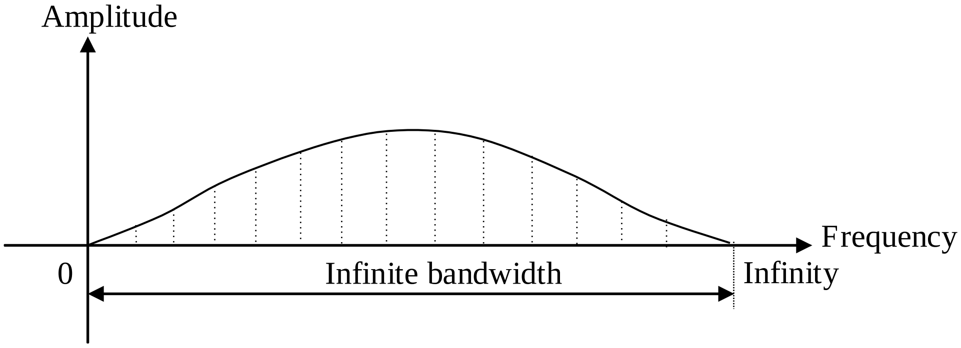


c) First, third, fifth, and seventh harmonics

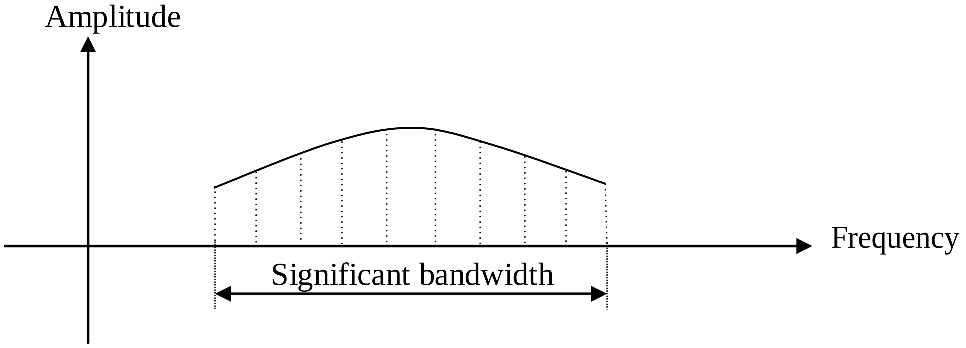


d) Infinite number of harmonics

If some of the components do not pass through the medium, this results in distortion of the signal at the receiver side. Since no practical medium (such as a cable) is capable of transferring the entire range of frequencies, there will always be distortion.



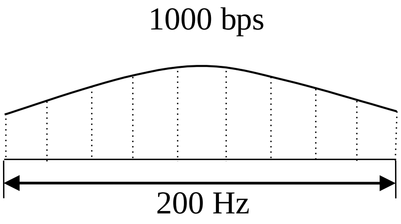
a) Spectrum for exact replica

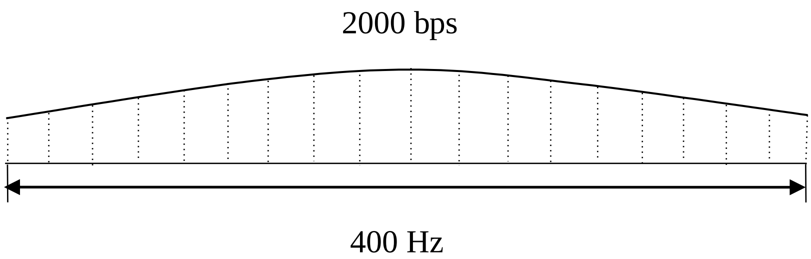


b) Significant spectrum

The part of the infinite spectrum whose amplitudes are significant (above an acceptable threshold), is called the significant spectrum, and its bandwidth is called the significant bandwidth.

When the bit rate increases, the significant bandwidth widens. For example, if the bit rate is 1000 bps, the significant bandwidth can be around 200 Hz, depending on the level of noise in the system. If the bit rate is 2000 bps, the significant bandwidth can be 400 Hz.

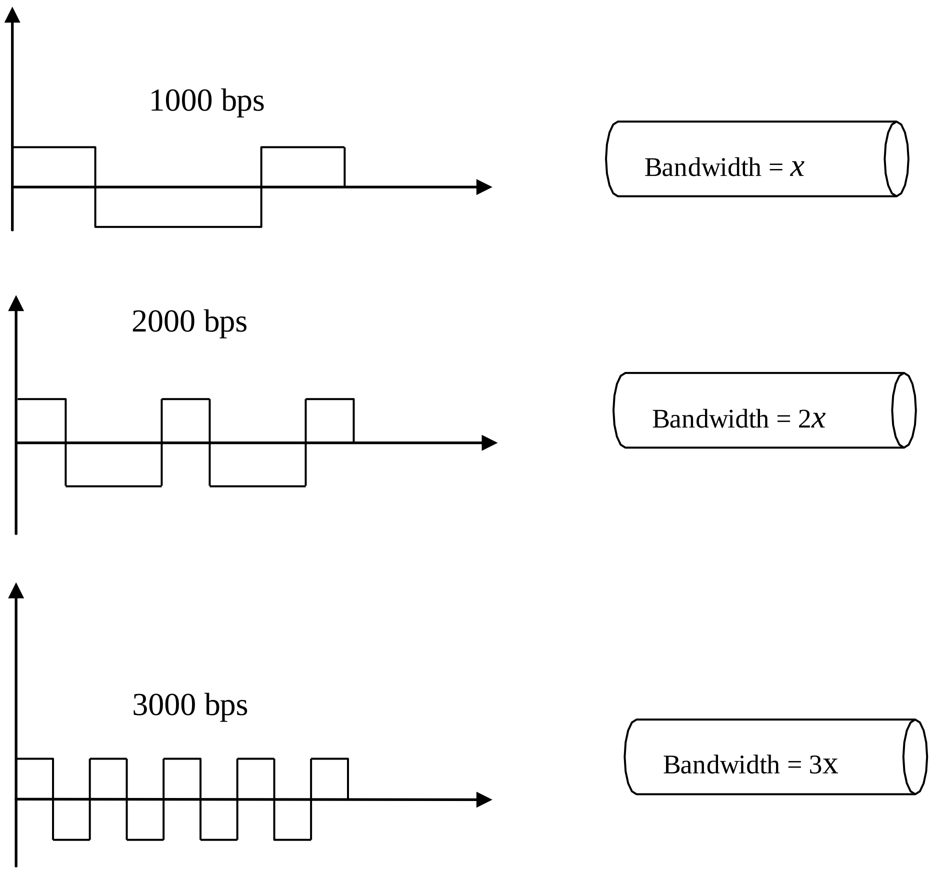




A transmission medium with a particular bandwidth is capable of transmitting only digital signals whose significant bandwidth is less than the bandwidth of the medium.

Channel Capacity

The maximum bit rate a transmission medium can transfer is called channel capacity of the medium. The capacity of a channel, however, depends on the type of encoding technique and the signal-to-noise ratio of the system. For example a normal telephone line with a bandwidth of 3000 Hz is capable of transferring up to 20,000 bps, but other factors, like noise, can decrease this rate.



Noise

In the absence of a signal, a transmission medium ideally has no electrical signal present. In practice, however, there is what we call line noise level, because of random perturbations on the line even when no signal is being transmitted. An important parameter associated with a transmission medium, therefore, is the ratio of the average power in a received signal, S, to the power in the noise level, N. The ratio S/N is known as the **signal-to-noise ratio (SNR)** and normally is expressed in decibels, that is:

$$\text{SNR} = 10\log_{10}\left(\frac{S}{N}\right) dB$$

- A high SNR ratio means a good-quality signal.

- A low SNR ratio means a low-quality signal.

The theoretical maximum data rate of transmission channel is related to the SNR ratio and we can determine this rate using a formula attributed to Shannon and Hartley. This is known as the **Shannon-Hartley Law**, which states:

$$C = W\log_2\left(1+\frac{S}{N}\right)\text{bps}$$

$$\approx 3.32\,W\log_{10}\left(1+\frac{S}{N}\right)\text{bps}$$

where C is the data rate in bps, W is the bandwidth of the line channel in Hz, S is the average signal power in watts and N is the random noise power in watts.

Ex

Consider a voice channel with BW of 2,800 Hz. A typical value of S/N for a telephone line is 20 dB. What is the channel capacity?

Solution

SNR = 20 dB

$20 = 10\log_{10}(S/N) \Rightarrow S/N = 100$

W = 2,800 Hz

$$C = W\log_2\left(1+\frac{S}{N}\right)\text{bps} \quad \approx 3.32\,W\log_{10}\left(1+\frac{S}{N}\right)\text{bps}$$

$$\approx 3.32\,(2800)\log_{10}(1+100)$$

$$C = 18{,}632\text{ bps}$$

CHAPTER 3

TRANSMISSION MEDIA

There are two basic categories of transmission media: guided and unguided media.

Guided transmission media use cabling system that guides the data signals along a specific path. Data signals are bound by the cabling system. Guided media is also known as bound media. ―Cabling‖ is meant in a generic sense, and is not meant to be interpreted as copper wire cabling only.

Unguided transmission media consists of a means for the data signals to travel but nothing to guide them along a specific path. The data signals are not bound to a cabling media and are therefore often called unbound media.

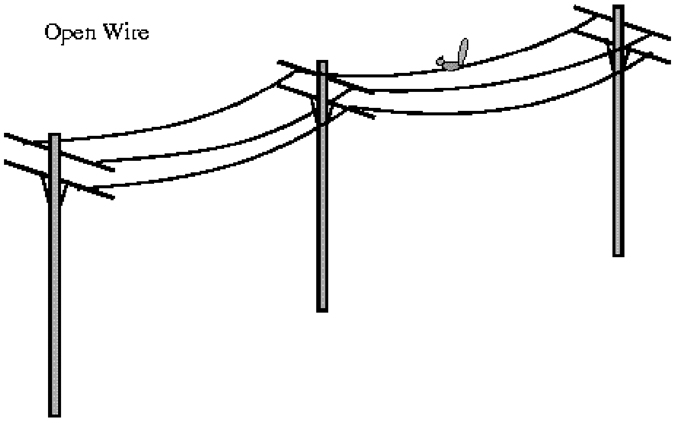
Transmission Media: Guided

There four basic types of guided media:

a. Open Wire
b. Twisted Pair
c. Coaxial Cable
d. Optical Fibre

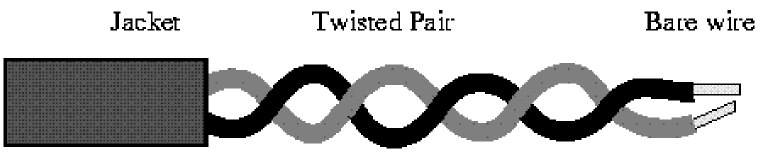
Open Wire

Open wire is traditionally used to describe the electrical wire strung along power poles. There is a single wire strung between poles. No shielding or protection from noise interference is used. We are going to extend the traditional definition of open wire to include any data signal path without shielding or protection from noise interference. This can include multi conductor cables or single wires. This medium is susceptible to a large degree of noise and interference and consequently is not acceptable for data transmission except for short distances under 20 ft.



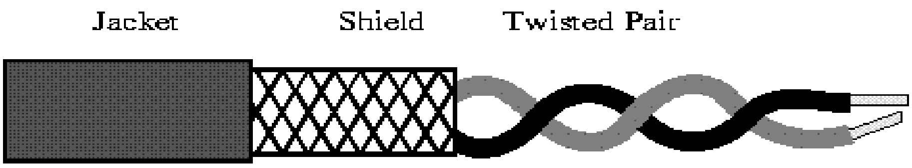
Twisted Pair

The wires in twisted pair cabling are twisted together in pairs. Each pair consists of a wire used for the positive data signal and a wire used for the negative data signal. Any noise that appears on one wire of the pair will also occur on the other wire. Since the wires have opposite polarities, they are 180 degrees out of phase. When noise appears on both wires, it cancels or nulls itself out at the receiving end. Twisted pair cables are most effectively used in systems that use a balanced line method of transmission: polar line coding (Manchester encoding) as opposed to unipolar line coding.



Unshielded Twisted Pair

The degree of reduction in noise interference is determined specifically by the number of turns per foot. Increasing the number of turns per foot reduces the noise interference. To further improve noise rejection, a foil or wire braid –shield‖ is woven around the twisted pairs. This shield can be woven around individual pairs or around a multi-pair conductor (several pairs).



Shielded Twisted Pair

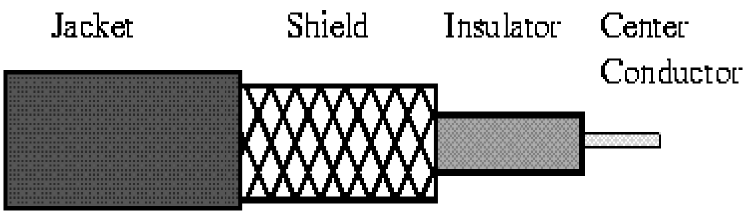
Cables with a shield are called shielded twisted pair and are commonly abbreviated STP. Cables without a shield are called unshielded twisted pair or UTP. Twisting the wires together results in a characteristic impedance for the cable. Typical impedance for UTP is 100 Ohm for Ethernet 10BaseT cable.

UTP or unshielded twisted pair cable is used in Ethernet 10BaseT and can also be used with Token Ring. It uses the RJ line of connectors (RJ45, RJ11, etc..).

STP or shielded twisted pair is used with the traditional Token Ring cabling or ICS-IBM Cabling System. It requires a custom connector. IBM STP (shielded twisted pair) has a characteristic impedance of 150 Ohm.

Coaxial Cable

Coaxial cable consists of two conductors. The inner conductor is held inside an insulator with the other conductor woven around it providing a shield. An insulating protective coating called a jacket covers the outer conductor.



Coaxial Cable

The outer shield protects the inner conductor from outside electrical signals. The distance between the outer conductor (shield) and inner conductor, plus the type of material used for insulating the inner conductor determine the cable properties or impedance. Typical impedances for coaxial cables are 75 Ohms for TV cable, 50 Ohms for Ethernet Thinnet and Thicknet. The excellent control of the impedance characteristics of the cable allow higher data rates to be transferred than with twisted pair cable.

Optical fibre

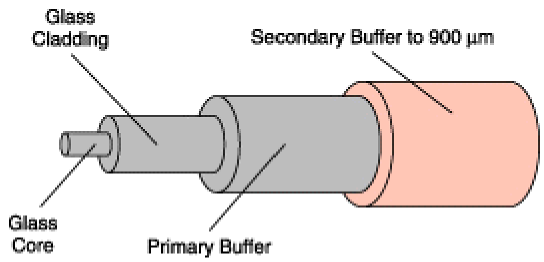
Optical fibre consists of thin glass fibres that can carry information at frequencies in the visible light spectrum and beyond. The typical optical fibre consists of a very narrow strand of glass called the core. Around the core is a concentric layer of glass called the cladding. A

typical core diameter is 62.5 microns (1 micron = 10^{-6} m). Typically Cladding has a diameter of 125 microns. Coating the cladding is a protective coating consisting of plastic, it is called the Jacket.

Fibre Optic Cables

Just as standard electric cables come in a variety of sizes, shapes, and types, fibre optic cables are available in different configurations. The simplest cable is just a single strand of fibre, whereas complex cables are made up of multiple fibres with different layers and other elements.

The portion of a fibre optic cable (core) that carries the light is made from either glass or plastic. Another name for glass is silica. Special techniques have been developed to create nearly perfect optical glass or plastic, which is transparent to light. Such materials can carry light over a long distance. Glass has superior optical characteristics over plastic. However, glass is far more expensive and more fragile than plastic. Although the plastic is less expensive and more flexible, its attenuation of light is greater. For a given intensity, light will travel a greater distance in glass than in plastic. For very long distance transmission, glass is certainly preferred. For shorter distances, plastic is much more practical.



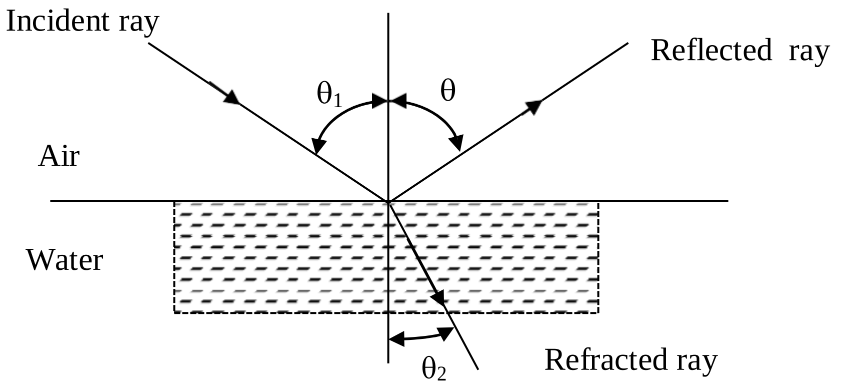
All fibres consist of a number of substructures including:

A core, which carries most of the light, surrounded by
A cladding, which bends the light and confines it to the core, surrounded by
A substrate layer (in some fibres) of glass which does not carry light, but adds to the diameter and strength of the fibre, covered by
A primary buffer coating, this provides the first layer of mechanical protection, covered by
A secondary buffer coating, this protects the relatively fragile primary coating.

The cladding is also made of glass or plastic but has a lower index of refraction. This ensures that the proper interface is achieved so that the light waves remain within the core. In addition to protecting the fibre core from nicks and scratches, the cladding adds strength. Some fibre optic cables have a glass core with a glass cladding. Others have a plastic core with a plastic cladding. Another common arrangement is a glass core with a plastic cladding. It is called plastic-clad silica (PCS) cable.

An important characteristic of fibre optics is refraction. Refraction is the characteristic of a material to either pass or reflect light. When light passes through a medium, it "bends" as it passes from one medium to the other. An example of this is when we look into a pond of water.

In 1621, the Dutch mathematician Willebrard Snell established that rays of light can be traced as they propagate from one medium to another based on their indices of refraction. Snell's low is stated by the equation:



$$\frac{n_1}{n_2} = \frac{\sin\theta_2}{\sin\theta_1};$$

$$n_1 \sin\theta_1 = n_2 \sin\theta_2$$

where n_1-refractive index of material 1; θ_1-angle of incidence; θ_2-angle of refraction; n_2-refractive index of material 2. When the angle of incidence, θ_1, becomes large enough to cause the sine of the refraction angle, θ_2, to exceed the value of 1, total internal reflection occurs. This angle is called the critical angle, θ_c. The critical angle, θ_c, can be derived from Snell's law as follows

$$n_1 \sin\theta_1 = n_2 \sin\theta_2$$

$$\sin\theta_1 = n_2 \sin\theta_2/n_1$$

When $\sin\theta_1 = \sin\theta_2$, then $\sin\theta_1 = n_2 / n_1$. Therefore, the critical angle: $\theta_c = \sin^{-1}(n_2 / n_1)$
Its index of refraction, however, it is typically 1% less than that of its core. This permits total internal reflection of rays entering the fibre and striking the core-cladding interface above the critical angle of approximately 82-degree ($\sin^{-1}$ (1/1.01). The core of the fibre therefore guides the light and the cladding contains the light. The cladding material is much less transparent than the glass making up the core of the fibre. This causes light rays to be absorbed if they strike the core-cladding interface at an angle less than the critical angle.

If the angle of incidence is small, the light rays are reflected and do not pass into the water. If the angle of incident is great, light passes through the media but is bent or refracted.

In the following figure, a light ray is transmitted into the core of an optical fibre. Total

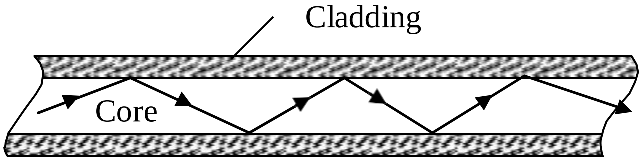


Figure 2

internal reflection occurs as it strikes the lower index cladding material.

Optical fibres work on the principle that the core refracts the light and the cladding reflects the light. The core refracts the light and guides the light along its path. The cladding reflects any light back into the core and stops light from escaping through it.

Transmission Modes

There are three primary types of transmission modes using optical fibre. They are

a. Step Index
b. Graded Index
c. Single Mode

Step index has a large core, so the light rays tend to bounce around inside the core, reflecting off the cladding. This causes some rays to take a longer or shorter path through the core. Some take the direct path with hardly any reflections while others bounce back and forth

taking a longer path. The result is that the light rays arrive at the receiver at different times. The signal becomes longer than the original signal. LED light sources are used. Typical Core: 62.5 microns.

Graded index has a gradual change in the core's refractive index. This causes the light rays to be gradually bent back into the core path. This is represented by a curved reflective path in the attached drawing. The result is a better receive signal than with step index. LED light sources are used. Typical Core: 62.5 microns.

Note: Both step index and graded index allow more than one light source to be used (different colours simultaneously), so multiple channels of data can be run at the same time!

Single mode has separate distinct refractive indexes for the cladding and core. The light ray passes through the core with relatively few reflections off the cladding. Single mode is used for a single source of light (one colour) operation. It requires a laser and the core is very small: 9 microns.

Basic Construction of Fibre-Optic Cables

There are two basic ways of classifying fibre optic cables. The first way is an indication of how the index of refraction varies across the cross section of the cable. The second way of classification is by mode. Mode refers to the various paths that the light rays can take in passing through the fibre. Usually these two methods of classification are combined to define the types of cable. There are two basic ways of defining the index of refraction variation across a cable. These are step index and graded index. Step index refers to the fact that there is a sharply defined step in the index of refraction where the fibre core and the cladding interface. It means that the core has one constant index of refraction n_1, while the cladding has another constant index of refraction n_2.

The other type of cable has a graded index. In this type of cable, the index of refraction of the core is not constant. Instead, the index of refraction varies smoothly and continuously over the diameter of the core. As you get closer to the centre of the core, the index of refraction gradually increases, reaching a peak at the centre and then declining as the other outer edge of the core is reached. The index of refraction of the cladding is constant.

Mode refers to the number of paths for the light rays in the cable. There are two classifications: single mode and multimode. In single mode, light follows a single path through the core. In multimode, the light takes many paths through the core.

Each type of fibre optic cable is classified by one of these methods of rating the index or mode. In practice, there are three commonly used types of fibre optic cable: multimode step index, single mode step index and multimode graded index cables.

1. Multimode Step-Index Fibre.

This cable (see Figure 5 (a)) is the most common and widely used type. It is also the easiest to make and, therefore, the least expensive. It is widely used for short to medium distances at relatively low pulse frequencies.

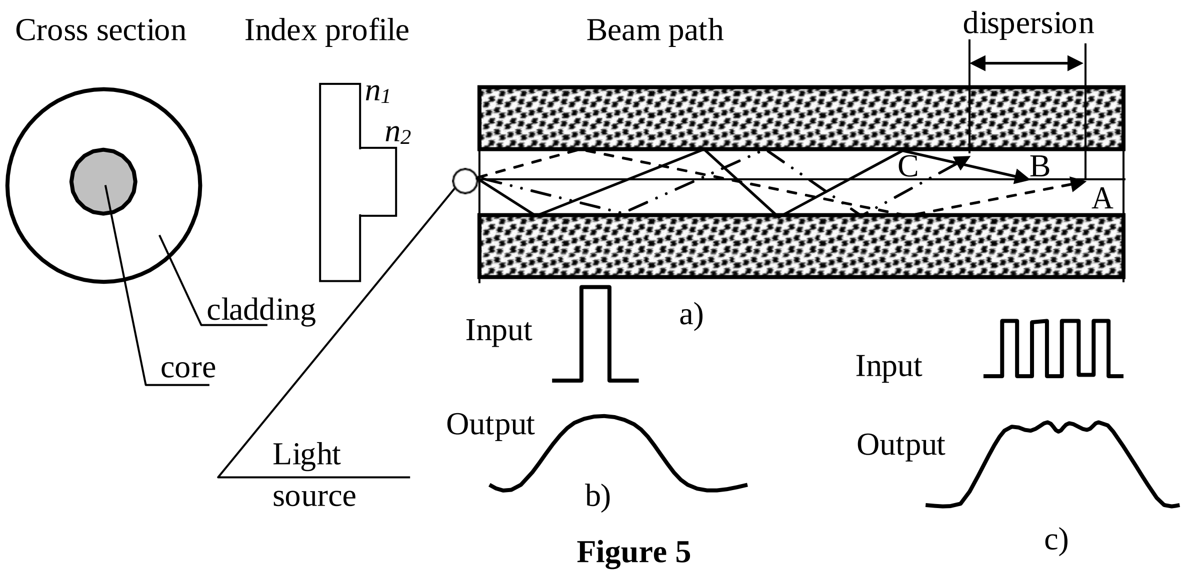


Figure 5

The main advantage of a multimode step index fibre is the large size. Typical core diameters are in the 50-to-1000 micrometers (μm) range. Such large diameter cores are excellent at gathering light and transmitting it efficiently. This means that an inexpensive light source such as LED can be used to produce the light pulses. The light takes many hundreds of even thousands of paths through the core before exiting. Because of the different lengths of these paths, some of the light rays take longer to reach the end of the cable than others. The problem with this is that it stretches the light pulses (Figure 5 (b). In Figure 5 ray A reaches the end first, then B, and C. The result is a pulse at the other end of the cable that is lower in amplitude due to the attenuation of the light in the cable and increased in duration due to the different arrival times of the various light rays. The stretching of the pulse is referred to as modal dispersion. Because the pulse has been stretched, input pulses can not occur at a rate faster than the output pulse duration permits. Otherwise the pulses will essentially merge together as shown in Figure 5 (c). At the output, one long pulse will occur and will be indistinguishable from the three separate pulses originally transmitted. This means that incorrect information will be received. The only core for this problem is to reduce the pulse

repetition rate. When this is done, proper operation occurs. But with pulses at a lower frequency, less information can be handled.

2. Single Mode Cable

In a single mode, or mono-mode, step-index fibre cable the core is so small that the total number modes or paths through the core are minimised and modal dispersion is essentially eliminated. The typical core sizes are 5 to 15 μm. The output pulse has essentially the same duration as the input pulse (see Figure 6).

The single mode step index fibres are by far the best since the pulse repetition rate can be high and the maximum amount of information can be carried. For very long distance transmission and maximum information content, single-mode step-index fibre cables should be used.

The main problem with this type of cable is that because of its extremely small size, it is difficult to make and is, therefore, very expensive. Handling, splicing, and making interconnections are also more difficult. Finally, for proper operation an expensive, super intense light source such as a laser must be used. For long distances, however, this is the type of cable preferred.

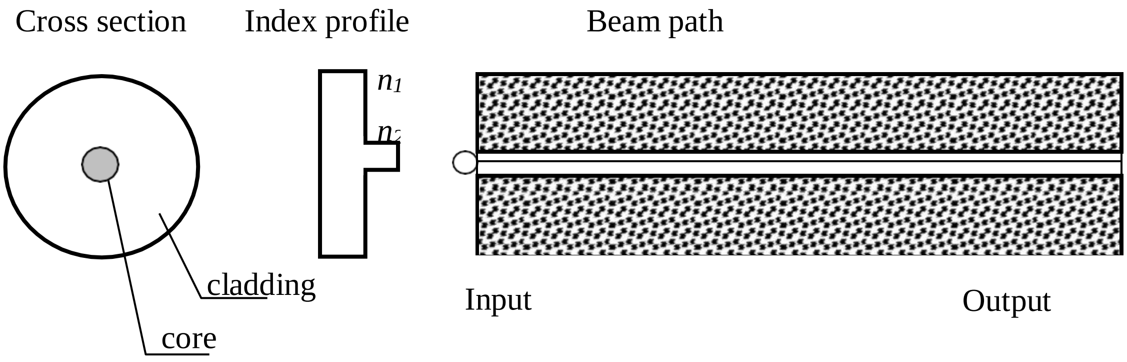


Figure 6

3. Multimode graded-index fibre cables

These cables have a several modes or paths of transmission through the cable, but they are much more orderly and predictable. Figure 7 shows the typical paths of the light beams. Because of the continuously varying index of refraction across the core, the light rays are bent smoothly and converge repeatedly at points along the cable.

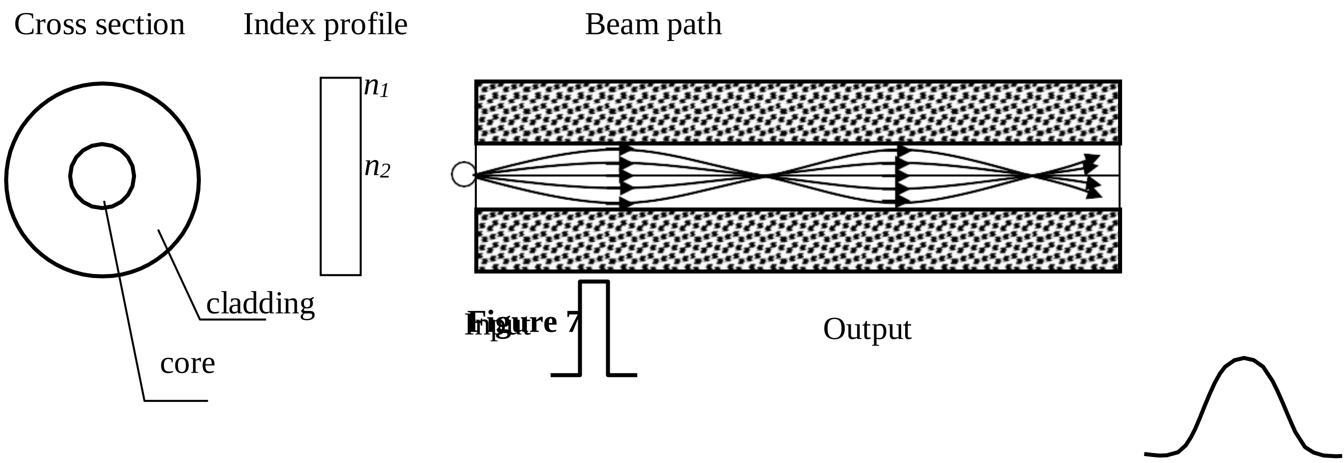


Figure 7

The light rays near the edge of the core take a longer path but travel faster since the index of refraction is lower. All the modes or light paths tend to arrive at one point simultaneously. The result is that there is less modal dispersion. It is not eliminated entirely, but the output pulse is not nearly as stretched as in multimode step index cable. The output pulse is only slightly elongated. As a result, this cable can be used at very high pulse rates and, therefore, a considerable amount of information can be carried on it.

This type of cable is also much wider in diameter with core sizes in the 50 to 100 (μm) range. Therefore, it is easier to splice and interconnect, cheaper, and less-intense light sources may be used. The most popular fibre-optic cables that are used in LAN: multimode-step index cable -65.5/125; multimode-graded index cable - 50/125. The multimode-graded index cable - 100/140 or 200/300 are recommended for industrial control applications because of its large size. In high data rate systems single mode fibre 9/125 is used. Typical core and cladding diameters of these cables are shown in Figure 8.

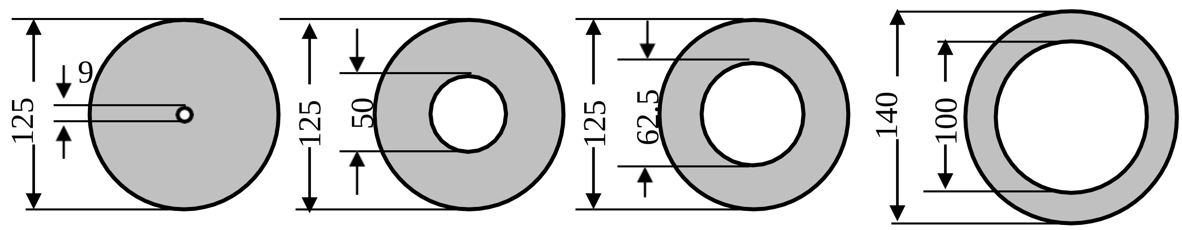


Figure 8

Specifications of the Fibre Cables

Indoor cable specifications:

- LED (Light Emitting Diode) light source
- 3.5 dB/Km Attenuation (loses 3.5 dB of signal per kilometer)

- 50 nM - wavelength of light source
- Typically 62.5/125 (core diameter/cladding diameter)
- Multimode - can run many light sources.

Outdoor cable specifications:

- Laser Light Source
- 1 dB/Km Attenuation (loses 1 dB of signal per kilometer)
- 1170 nM - wavelength of light source
- Monomode (single mode)

Advantages of Optical Fibre:

- Noise immunity: RFI and EMI immune (RFI - Radio Frequency Interference, EMI - Electromagnetic Interference)
- Security: cannot tap into cable.
- Large Capacity due to BW (bandwidth)
- No corrosion
- Longer distances than copper wire
- Smaller and lighter than copper wire
- Faster transmission rate

Disadvantages of optical fibre:

- Physical vibration will show up as signal noise!
- Limited physical arc of cable. Bend it too much and it will break!
- Difficult to splice

The cost of optical fibre is a trade-off between capacity and cost. At higher transmission capacity, it is cheaper than copper. At lower transmission capacity, it is more expensive.

Media versus Bandwidth

The following table compares the usable bandwidth of the different guided transmission media.

Cable Type	Bandwidth
Open Cable	0 - 5 MHz
Twisted Pair	0 - 100 MHz
Coaxial Cable	0 - 600 MHz
Optical Fibre	0 - 1 GHz

Transmission Media: Unguided

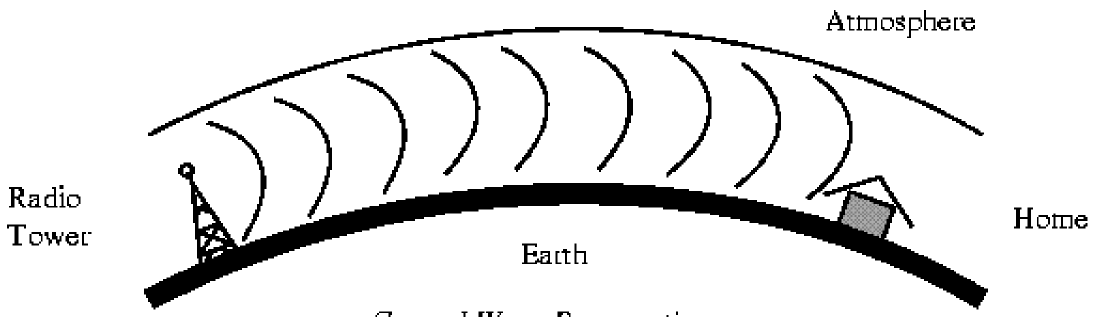
Unguided transmission media is data signals that flow through the air. They are not guided or bound to a channel to follow. They are classified by the type of wave propagation.

RF Propagation

There are three types of RF (radio frequency) propagation:

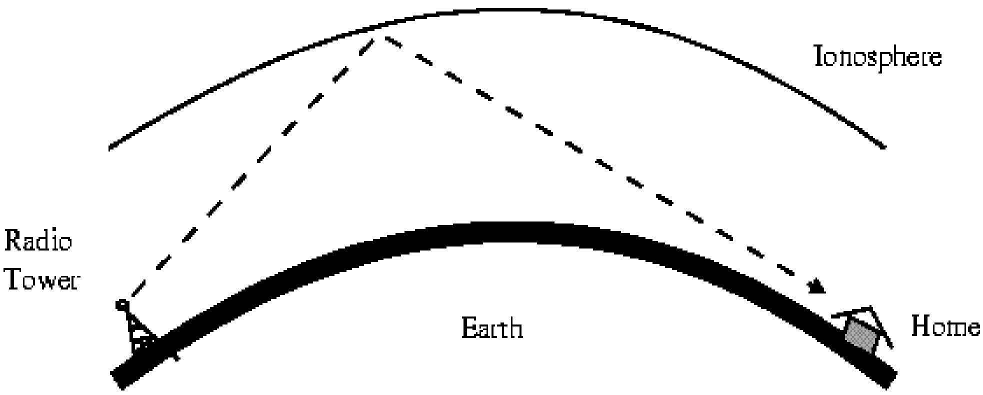
- Ground Wave
- Sky Wave
- Line of Sight (LOS)

Ground wave propagation follows the curvature of the Earth. Ground waves have carrier frequencies up to 2 MHz. AM radio is an example of ground wave propagation.

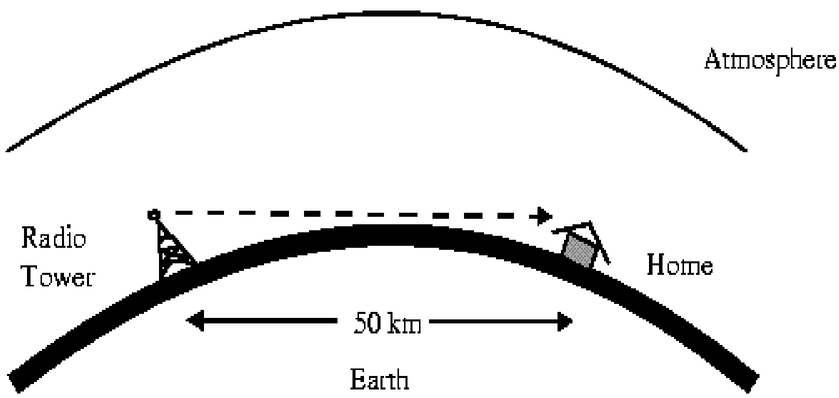


Ground Wave Propagation

Sky wave propagation bounces off of the Earth's ionospheric layer in the upper atmosphere. It is sometimes called double hop propagation. It operates in the frequency range of 30-85 MHz. Because it depends on the Earth's ionosphere, it changes with the weather and time of day. The signal bounces off of the ionosphere and back to earth. Ham radios operate in this range.



Line of sight propagation transmits exactly in the line of sight. The receive station must be in the view of the transmit station. It is sometimes called space waves or troposphere propagation. It is limited by the curvature of the Earth for ground-based stations (100 km, from horizon to horizon). Reflected waves can cause problems. Examples of line of sight propagation are: FM radio, microwave and satellite.



Radio Frequencies

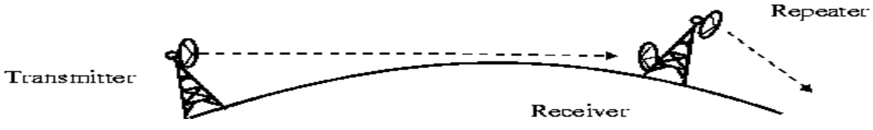
The frequency spectrum operates from 0 Hz (DC) to gamma rays (10^{19} Hz).

Name	Frequency (Hertz)	Examples
Gamma Rays	10^{19}	
X-Rays	10^{17}	
Ultra-Violet Light	7.5×10^{15}	
Visible Light	4.3×10^{14}	
Infrared Light	3×10^{11}	
EHF - Extremely High Frequencies	30 GHz (Giga = 10^9)	Radar
SHF - Super High Frequencies	3 GHz	Satellite & Microwaves
UHF - Ultra High Frequencies	300 MHz (Mega = 10^6)	UHF TV (Ch. 14-83)
VHF - Very High Frequencies	30 MHz	FM & TV (Ch2 - 13)
HF - High Frequencies	3 MHz2	Short Wave Radio
MF - Medium Frequencies	300 kHz (kilo = 10^3)	AM Radio
LF – Low Frequencies	30 kHz	Navigation
VLF - Very Low Frequencies	3 kHz	Submarine Communications
VF - Voice Frequencies	300 Hz	Audio
ELF - Extremely Low Frequencies	30 Hz	Power Transmission

Radio frequencies are in the range of 300 kHz to 10 GHz. We are seeing an emerging technology called wireless LANs. Some use radio frequencies to connect the workstations together, some use infrared technology.

Microwave

Microwave transmission is line of sight transmission. The transmit station must be in visible contact with the receive station. This sets a limit on the distance between stations depending on the local geography. Typically the line of sight due to the Earth's curvature is only 100 km to the horizon! Repeater stations must be placed so the data signal can hop, skip and jump across the country.



Microwaves operate at high operating frequencies of 3 to 10 GHz. This allows them to carry large quantities of data due to their large bandwidth.

Advantages:

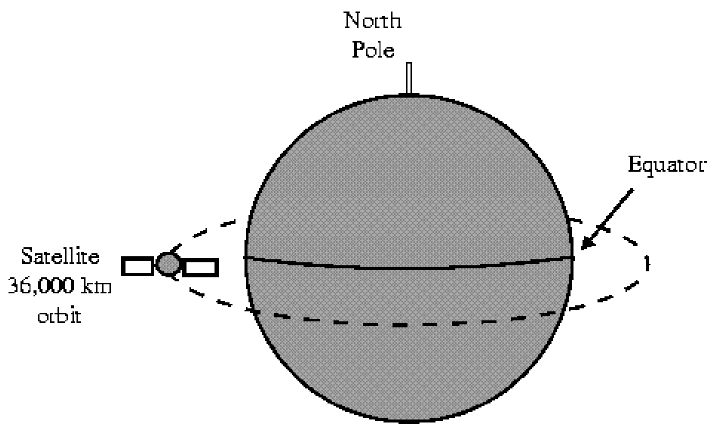
a. They require no right of way acquisition between towers.
b. They can carry high quantities of information due to their high operating frequencies.
c. Low cost land purchase: each tower occupies only a small area.
d. High frequency/short wavelength signals require small antennae.

Disadvantages:

a. Attenuation by solid objects: birds, rain, snow and fog.
b. Reflected from flat surfaces like water and metal.
c. Diffracted (split) around solid objects.
d. Refracted by atmosphere, thus causing beam to be projected away from receiver.

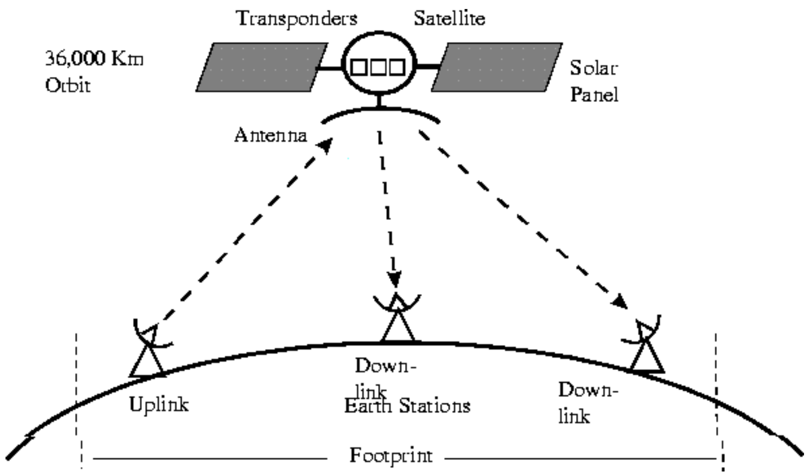
Satellite

Satellites are transponders (units that receive on one frequency and retransmit on another) that are set in geostationary orbits directly over the equator. These geostationary orbits are 36,000 km from the Earth's surface. At this point, the gravitational pull of the Earth and the centrifugal force of Earth's rotation are balanced and cancel each other out. Centrifugal force is the rotational force placed on the satellite that wants to fling it out into space.



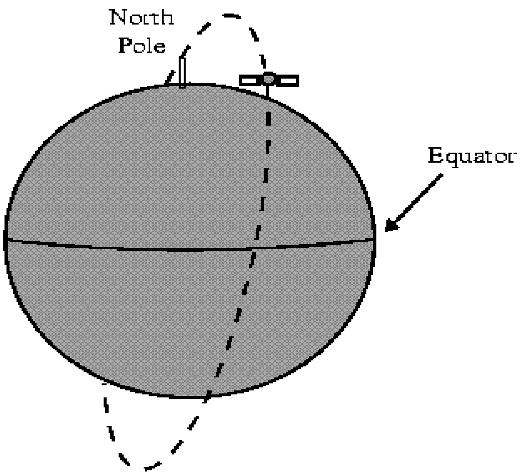
The uplink is the transmitter of data to the satellite. The downlink is the receiver of data. Uplinks and downlinks are also called Earth stations because they are located on the Earth.

The footprint is the "shadow" that the satellite can transmit to, the shadow being the area that can receive the satellite's transmitted signal.



Iridium Telecom System

The Iridium Telecom System is a new satellite system that will be the largest private aerospace project. It is a mobile telecom system intended to compete with cellular phones. It relies on satellites in lower Earth orbit (LEO). The satellites will orbit at an altitude of 900 - 10,000 km in a polar, non-stationary orbit. Sixty-six satellites are planned. The user's handset will require less power and will be cheaper than cellular phones. There will be 100% coverage of the Earth.



Unfortunately, although the Iridium project was planned for 1996-1998, with 1.5 million subscribers by end of the decade, it looked very financially unstable.

CHAPTER 4

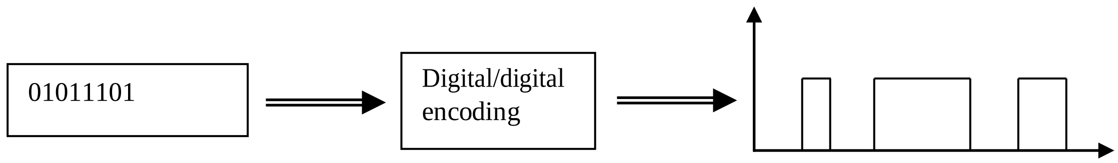
ENCODING, MODULATING & TRANSMISSION CODES

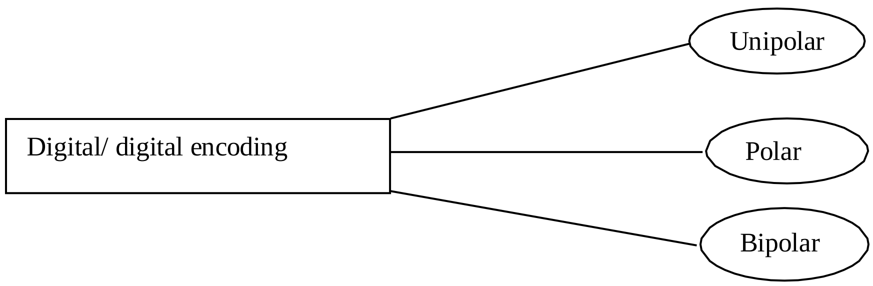
ENCODING (D/D) (A/D) (D/A) (A/A)

Information must be encoded into signals before it can be transported across communication media. We must encode data into signals to send them from one place to another.

Digital-to-Digital Encoding

Digital-to-Digital Encoding is the representation of digital information by a digital signal. (eg. computer-to-printer)



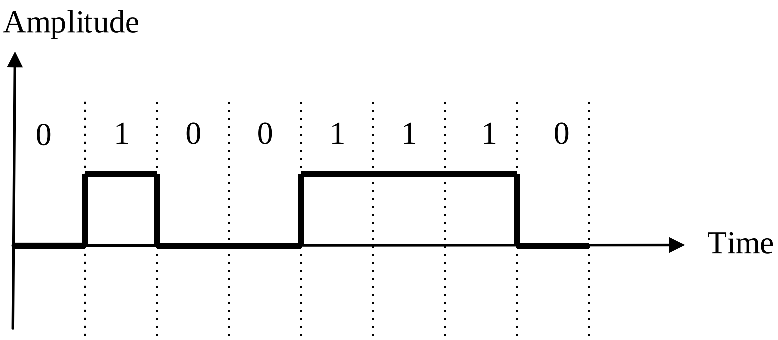


Unipolar

Digital transmission systems work by sending voltage pulses along a media link, usually a wire or a cable. In most types of encoding, one voltage level stands for binary 0 and another level stands for binary 1. The polarity of a pulse refers to whether it is positive or negative.

Unipolar encoding is so named because it uses only one polarity. Therefore, only one of the two binary states is encoded, usually the 1. The other state, usually 0, is represented by zero voltage, or an idle line.

Unipolar encoding uses only one level of value.



1's encoded as positive, 0's are idle. Unipolar encoding is straight forward and inexpensive to implement. However, it has two problems that make it unusable: DC component and synchronisation.

DC component

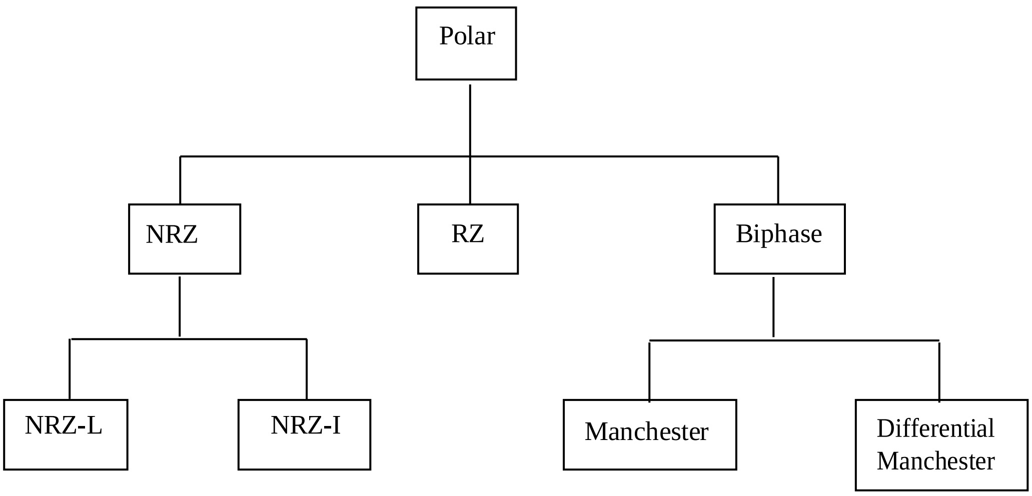
Average amplitude is nonzero → creates a direct current (DC) component, when a signal contains a DC component it cannot travel through media that cannot handle DC components: e.g. microwaves or transformers.

Synchronisation

When a signal is unvarying, the receiver cannot determine the beginning and ending of each bit. Therefore, Synchronisation problem in unipolar encoding can occur whenever the data stream includes a long uninterrupted series of 1's or 0's.

Polar Encoding

Polar encoding uses two voltage levels: one positive and one negative. In most polar encoding methods the average voltage level on the line is reduced and the DC component problem of unipolar encoding is alleviated.



Non-Return-to-Zero (NRZ) Encoding

In NRZ encoding, the level of the signal is always either positive or negative. In NRZ system if the line is idle it means no transmission is occurring at all.

- **NRZ-L (Non-return-to-zero, Level)**

  In NRZ-L the level of the signal is dependant upon the state of the bit.

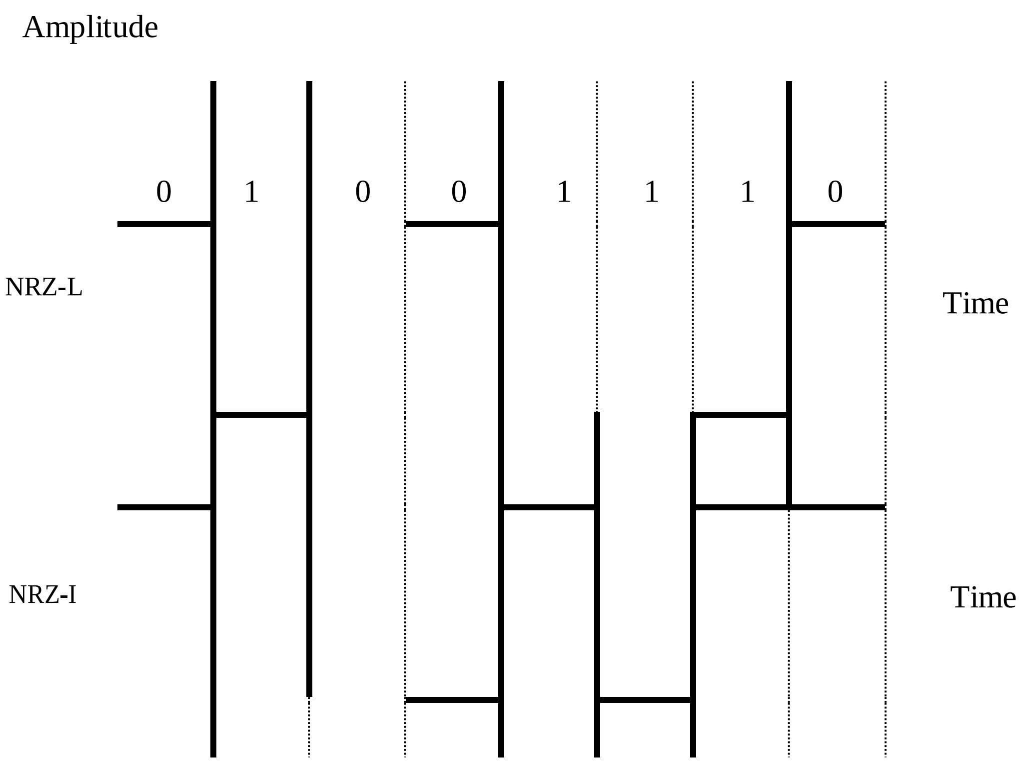
  A positive voltage usually means the bit is 0, and negative voltage means the bit is a 1 (and vice versa).

- **NRZ-I (Non-return-to-zero, Invert)**

  In NRZ-I, an inversion of the voltage level represents a 1 bit. It is the transition between a positive and a negative voltage, not the voltages themselves that represents a 1 bit. A 0 bit is represented by no change.

  An advantage of NRZ-I over NRZ-L is that because the signal changes every time a 1 bit is encountered, it provides some synchronisation.

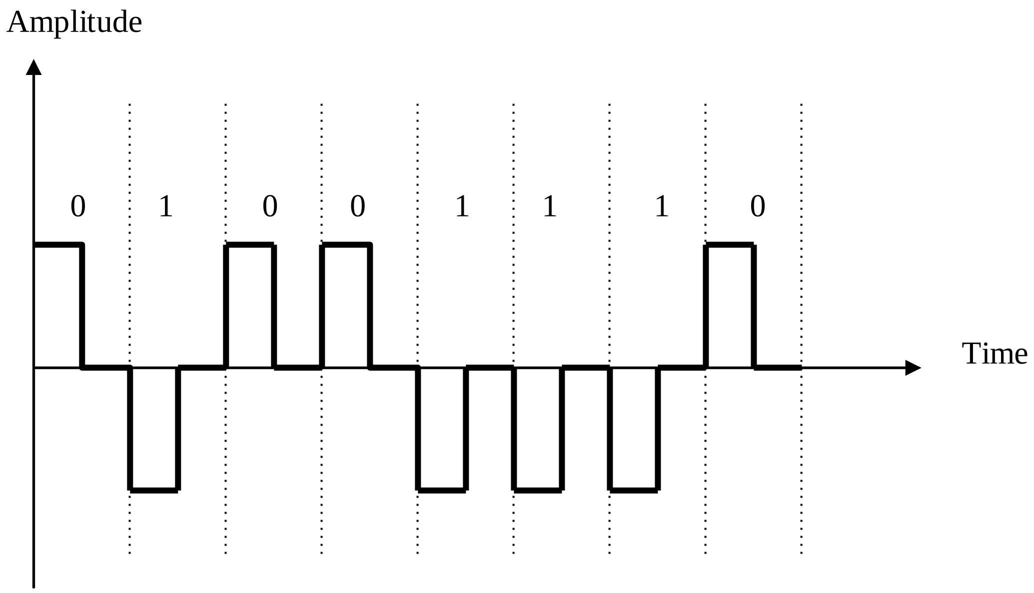
  Each inversion allows the receiver to synchronise its timer to the actual arrival of the transmission.



Z (Return-to-zero) Encoding

To assure synchronisation, there must be a signal change for each bit. The receiver can use these changes to built up, update, and synchronise its clock.

One solution is return to zero (RZ) encoding, which uses three Values: positive, negative, and zero.



The main disadvantage of RZ encoding is that it requires two signal changes to encode one bit and therefore occupies more bandwidth. But of the three alternatives discussed above, it is the most effective. Because a good encoded digital signal must contain a provision for synchronisation.

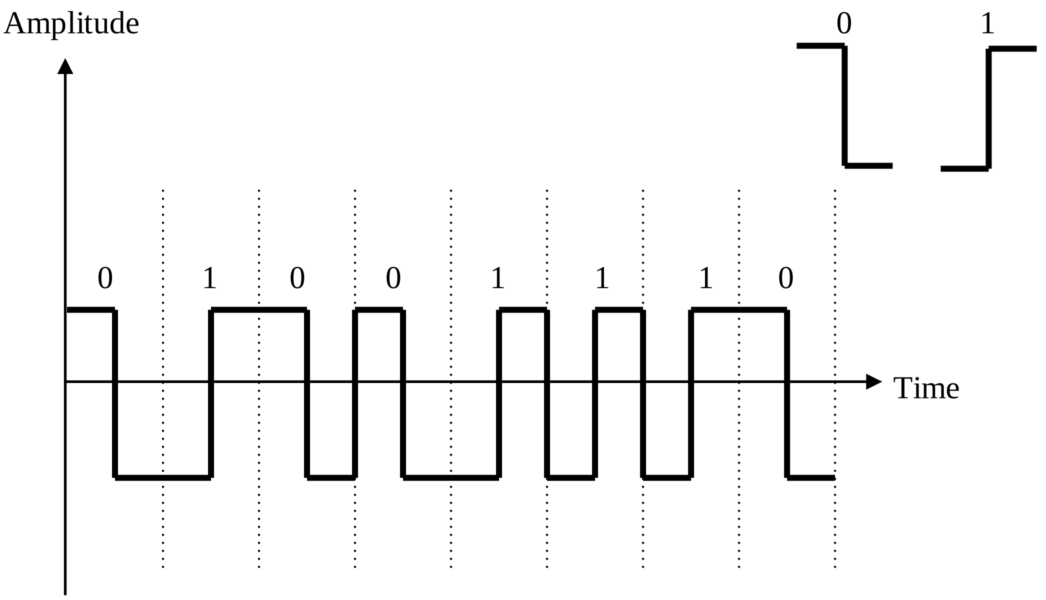
Biphase Encoding

Probably the best existing solution to the problem of synchronisation is biphase encoding. In this method, the signal changes at the middle of the bit interval but does not return to zero. Instead it continues to the opposite pole. As in RZ, these mid-interval transitions allow for synchronisation.

Biphase encoding is implemented in two different ways: Manchester and differential Manchester.

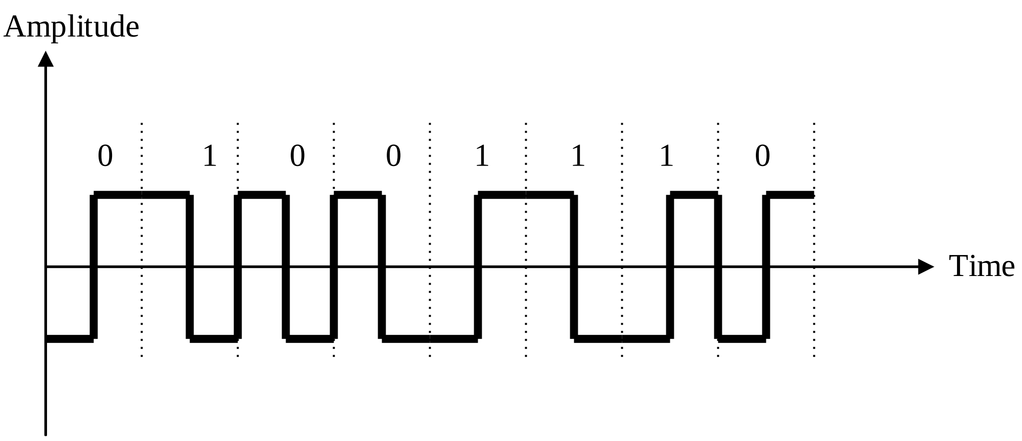
- **Manchester**

  Manchester encoding uses the inversion at the middle of each bit interval for both synchronisation and bit representation. A negative-to-positive transition represents binary 1 and a positive-to-negative transition represents binary 0.



- **Differential Manchester**

In this method, the inversion at the middle of the bit is used for synchronisation, but the presence or absence of an additional transition at the beginning of the interval is used to identify the bit. A transition means binary 0 and no transition means binary 1. The bit representation is shown by the inversion and non-inversion at the beginning of the bit.



Bipolar Encoding

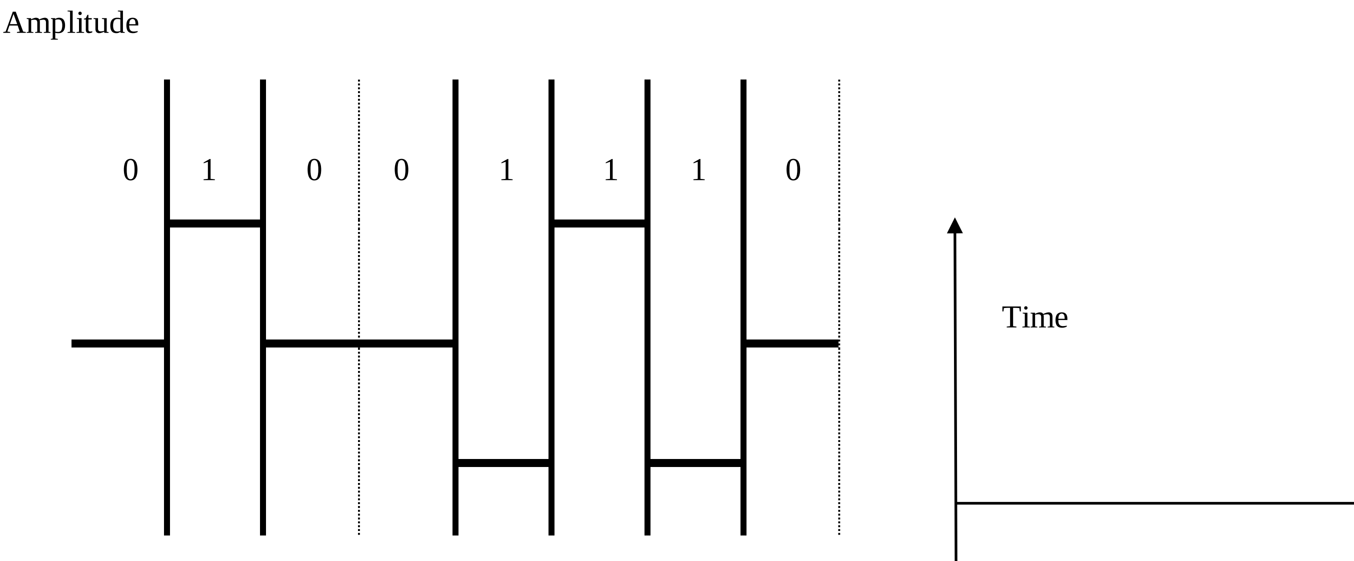
Bipolar encoding uses three voltage levels: positive, negative and zero. The zero level is used to represent binary 0 positive and negative voltages represent alternating 1s. (If 1^{st} one +ve, 2^{nd} is -ve).

* Three types of bipolar encoding are popular use by the data communications industry: AMI, B8ZS, and HDB3.

- **Bipolar Alternate Mark Inversion (AMI)**

  Bipolar AMI is the simplest type of bipolar encoding. The word mark comes from telegraphy and means 1.

  AMI means alternate 1 inversion. A neutral, zero voltage represents binary 0. Binary 1s are represented by alternating positive and negative voltages



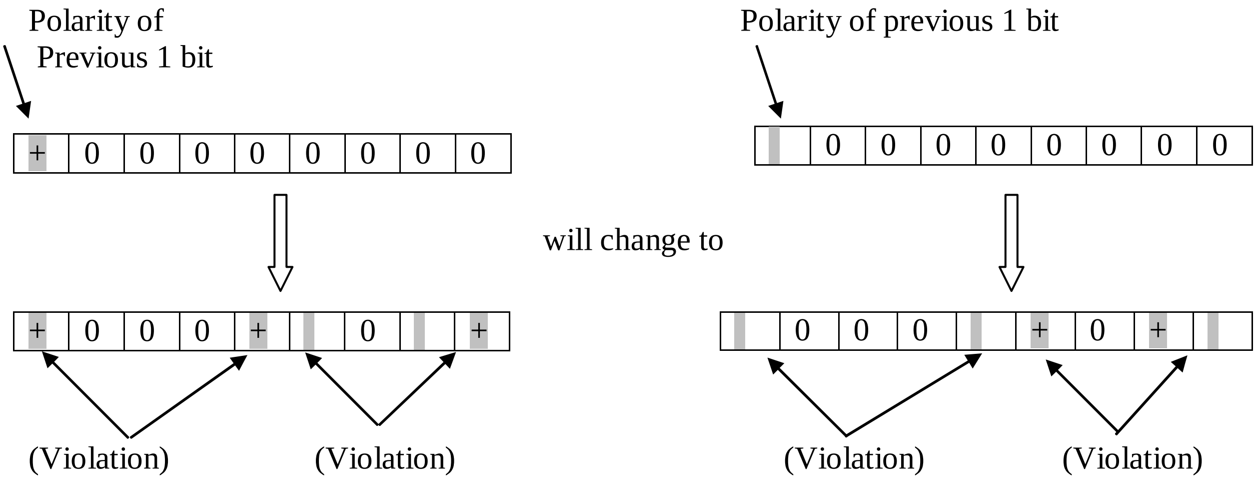
By inverting on each occurrence of a 1, bipolar AMI accomplishes two things: first, the DC component is zero, and second, a long sequence of 1s stays synchronised.

Two variations of bipolar AMI have been developed to solve the problem of synchronisation sequential 0s. The first used in North America, is called bipolar 8-zero substitution (B8ZS); the second, used in Europe and Japan, is called high-density bipolar 3 (HDB3). Both are adaptations of bipolar AMI that modify the original pattern only in the case of multiple consecutive 0s.

Bipolar 8-Zero Substitution (B8ZS)

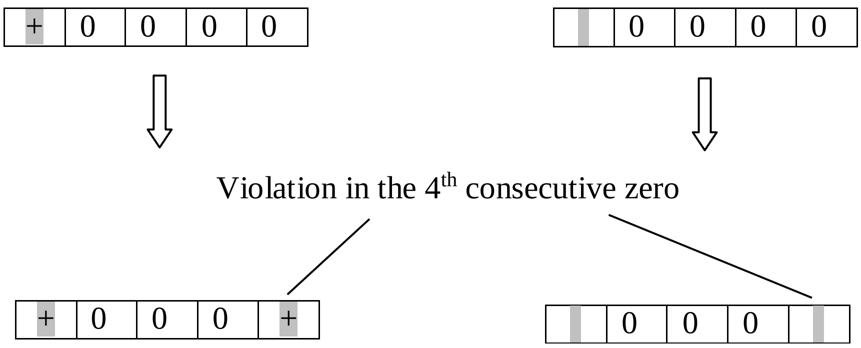
B8ZS is the convention adopted in North America to provide synchronisation of long strings of 0s. In most situations B8ZS functions identically to bipolar AMI. Bipolar AMI changes poles with every 1 it encounters. These changes provide the synchronisation needed by the receiver, but the signal does not change during a string of 0s, so synchronisation is lost. The solution provided by B8ZS is to force artificial signal changes, called violations

- In B8ZS, if eight 0s come one after another, we change the pattern in one of two ways based on the polarity of previous 1.

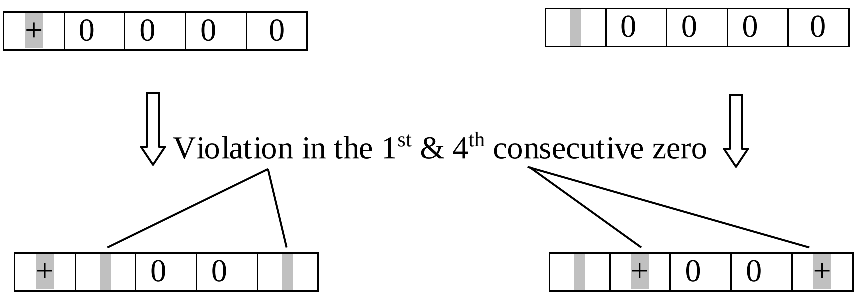


High-Density Bipolar 3 (HDB3)

In HDB3 if four 0s come one after another, we change the pattern in one of four ways based on the polarity of the previous 1 and the number of 1s since the last substitution.



If the number of 1s since the last substitution is odd



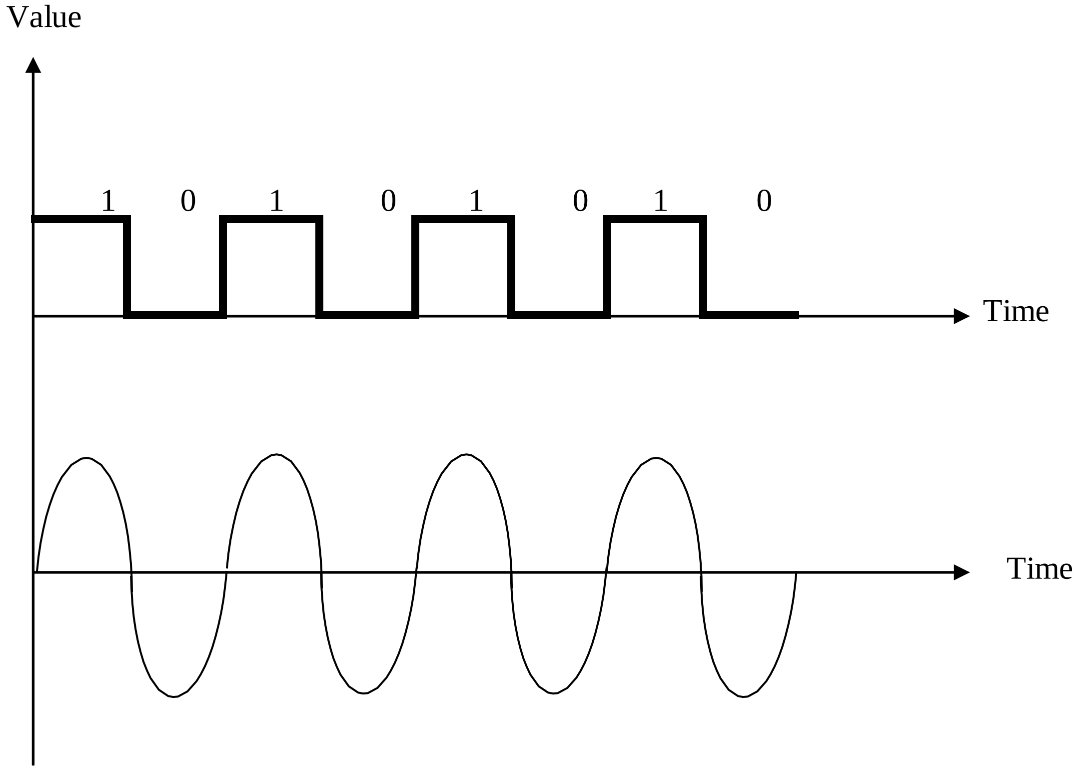
If the number of 1s since the last substitution is even

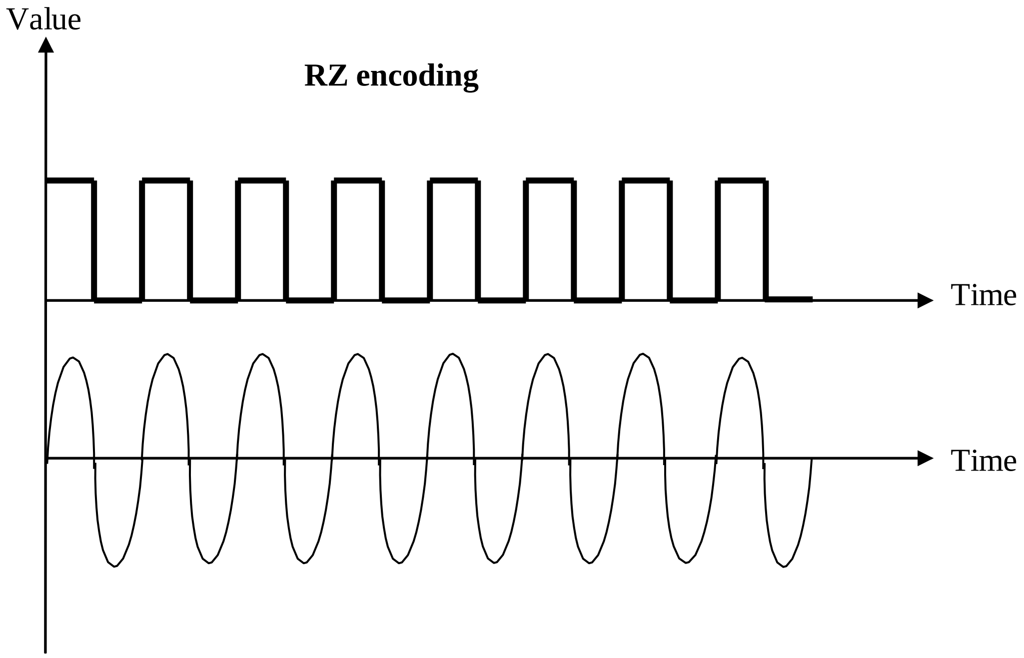
Compare the bandwidth needed for unipolar encoding and RZ encoding. Assume the worst-case scenario for both.

Solution

The worst case scenario (the situation requiring the most bandwidth) is alternating 1s and 0s for unipolar, for RZ the worst-case is all 1s.

Unipolar encoding





RZ needs twice the bandwidth of unipolar.

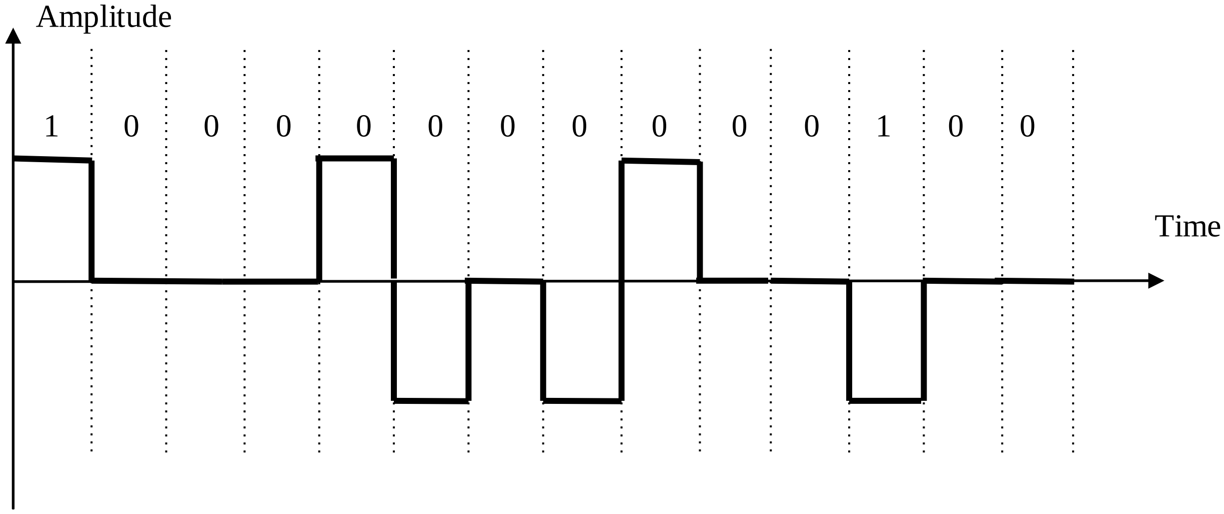
Compare the bandwidth needed for Manchester and Differential Manchester encoding. Assume the worst-case scenario for both.

Solution

The worst-case scenario for Manchester is consecutive 1s or consecutive 0s. There are two transistors for each bit (one cycle per bit). For Differential Manchester the worst – case is consecutive 0s with two transitions per each bit (one cycle per bit). The bandwidths, which are proportional to bit rate, are the same for each.

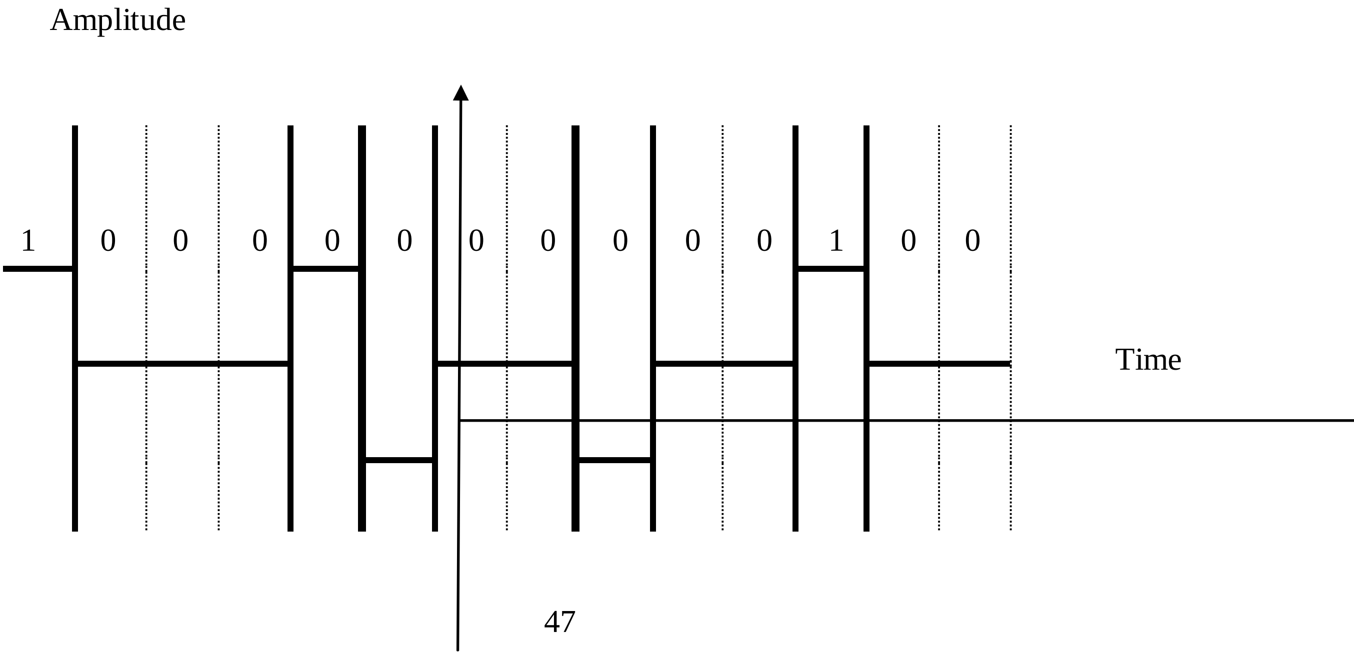
Ex

Using B8ZS, encode the bit stream 10000000000100. Assume that the polarity of the previous 1 is positive.

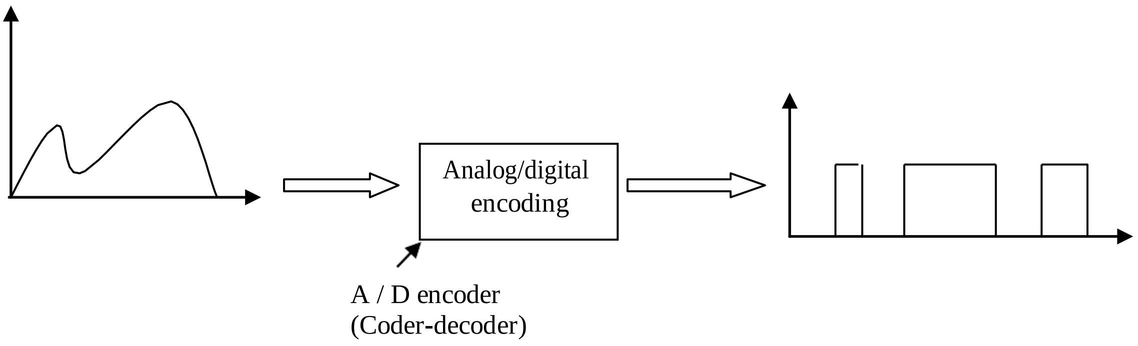


Ex

Using HDB3, encode 10000000000100. Assume that the number of 1s so far is odd and the previous 1 is positive.



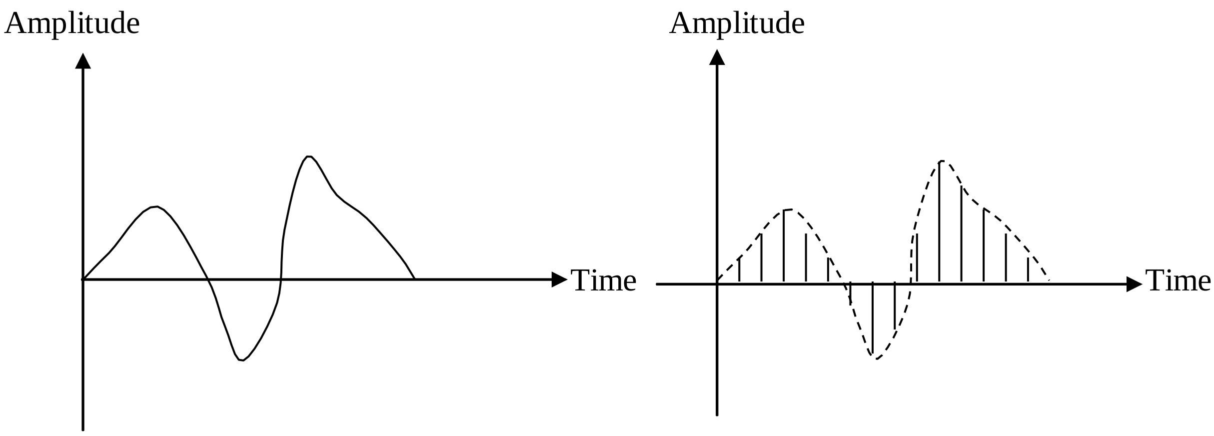
Analog-to- Digital Encoding



In analog-to-digital encoding, the information contained in a continuous wave form are represented as a series of digital pulses (1s and 0s).

Pulse Amplitude Modulation (PAM)

The first step in A/D encoding is called pulse amplitude modulation (PAM). This technique takes analog information, samples it, and generates a series of pulses based on the results of sampling. The term sampling means measuring the amplitude of the signal at equal time intervals.



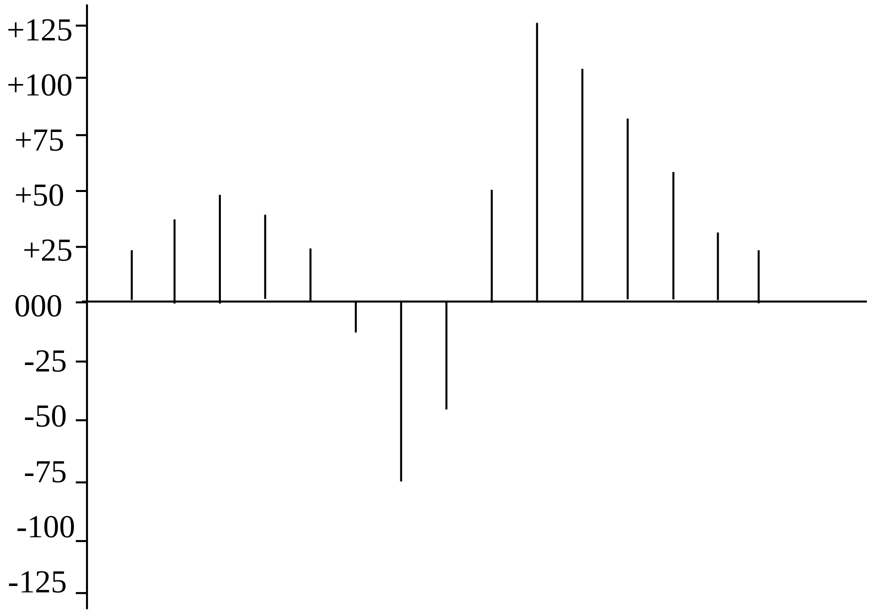
Analog Signal **PAM Signal**

In PAM, the original signal is sampled at equal intervals.

PAM has some applications, but it is not used by itself in data communications. However, it is the first step in another very popular encoding method called pulse code modulation (PCM).

Pulse Code Modulation (PCM)

PCM modifies the pulses created by PAM to create a complete digital signal. To do so, PCM first quantises the PAM pulses. Quantisation is a method of assigning integral values in a specific range to sampled instances. (The result of quantisation is presented in the following figure).

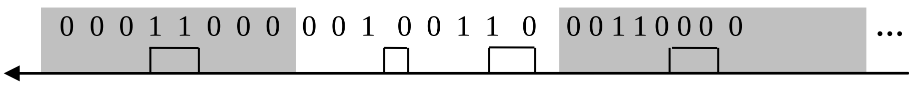


Each value is translated into its seven-bit binary equivalent. The eighth bit indicates the sign.

+24	00011000	-15	10001111	+125	01111101
+38	00100110	-80	11010000	+110	01101110
+48	00110000	-50	10110010	+90	01011010
+39	00100111	+52	00110110	+88	01011000
+26	00011010	+127	01111111	+77	01001101

The binary digits are then transformed into a digital signal using one of the digital encoding.

PCM



Direction of transfer

The result of the PCM of the original signal encoded finally into a unipolar signal.

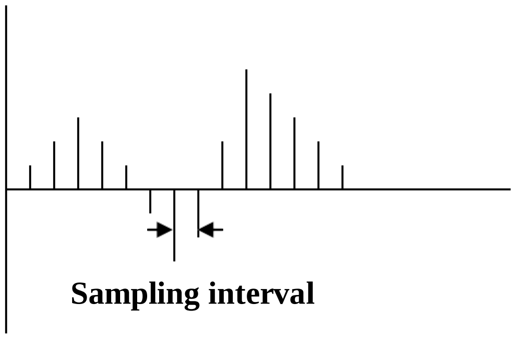
PCM is actually made up of four separate processes: PAM, quantisation, binary encoding, and digital-to-digital encoding.

PCM is the sampling method used to digitize voice in T-line transmission in the North America telecommunication system.

According to the Nyquist theorem, the sampling rate must be at least two times the highest frequency.

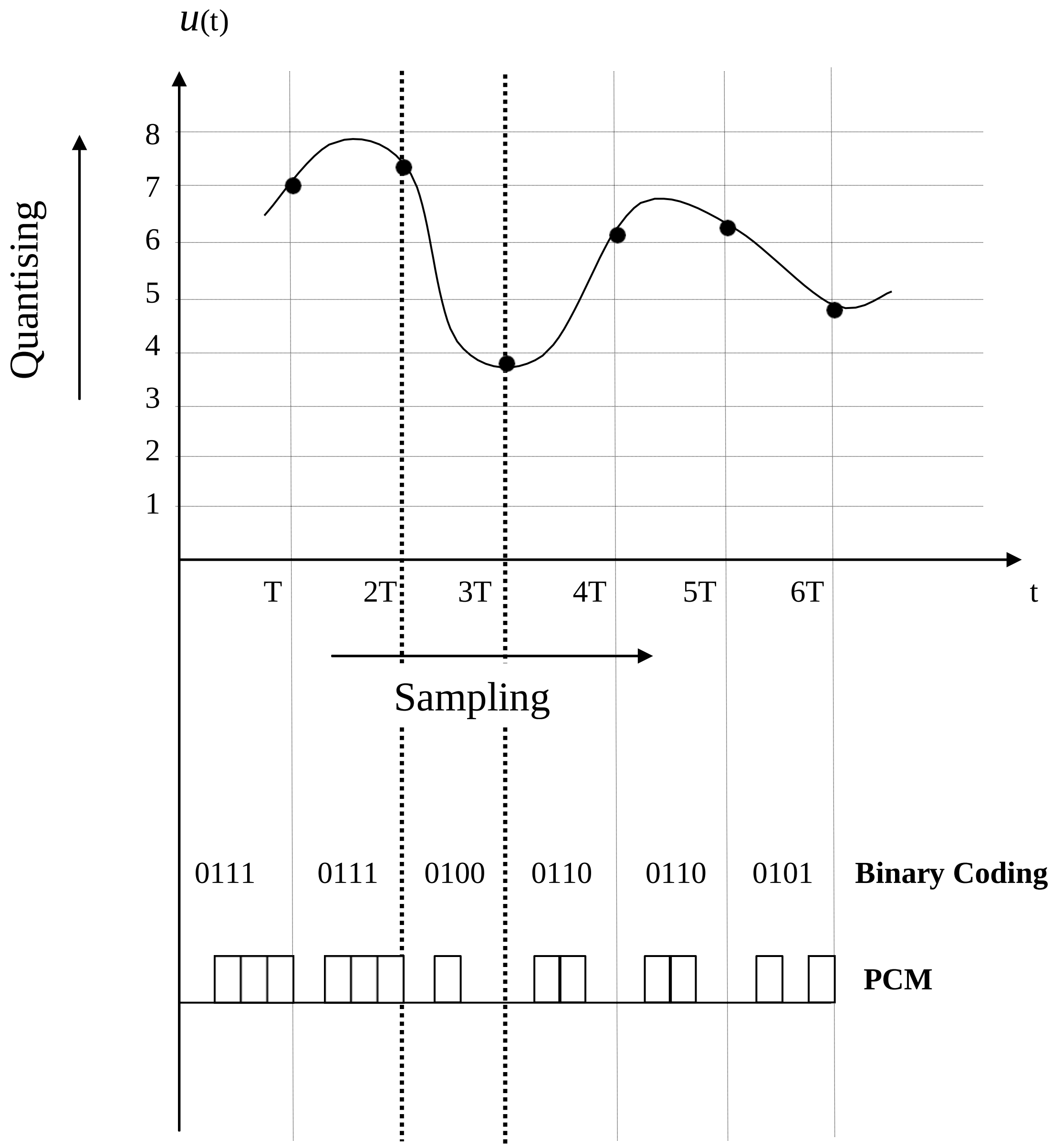
$$\text{Highest frequency} = x \text{ Hz}$$

$$\text{Sampling rate} = 2x \text{ samples/second}$$



Quantisation

Quantising is the process of rounding-off the values of the flat-top samples to certain predetermined levels.

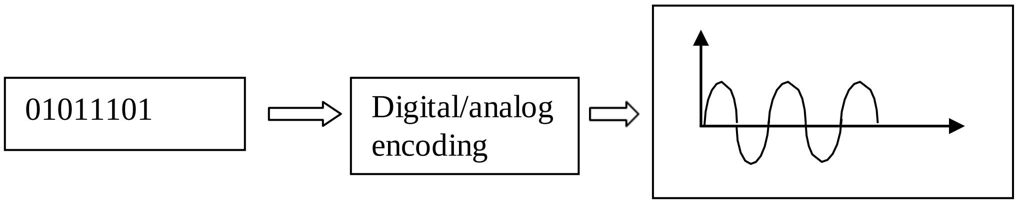


Ex What sampling rate is needed for a signal with a bandwidth of 10,000 Hz (1000 Hz to 11,000 Hz)? If the quantisation is eight bits per sample, what is the bit rate?

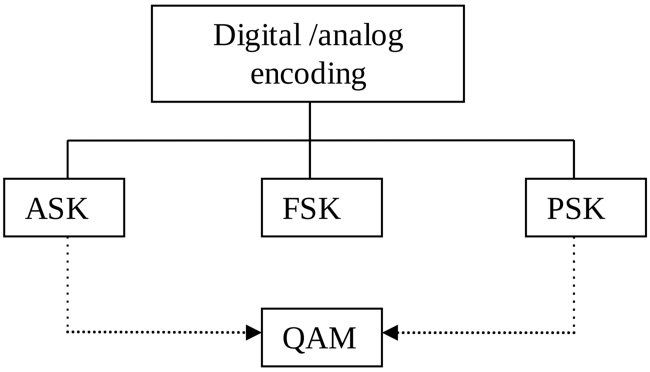
Solution

Sampling rate = 2 (11,000) = 22,000 samples/s each sample is quantised to eight bits: data rate = (22,000 samples/s) (8 bits/sample) = 176 kbps

Digital-to-Analog Encoding



Digital-to-analog encoding is the representation if digital information by an analog signal.



Quadrature Amplitude Modulation

Bit Rate and Baud Rate

- Bit rate is the number of bits transmitted in one second.
- Baud rate refers to the number of signal units per second that are required to represent those bits.
- For computer efficiency, bit rate is more important.
- For data transmission, baud rate is more important ⇨ the fewer the signal units required, the more efficient the system, and the less bandwidth required to transmit more bits.

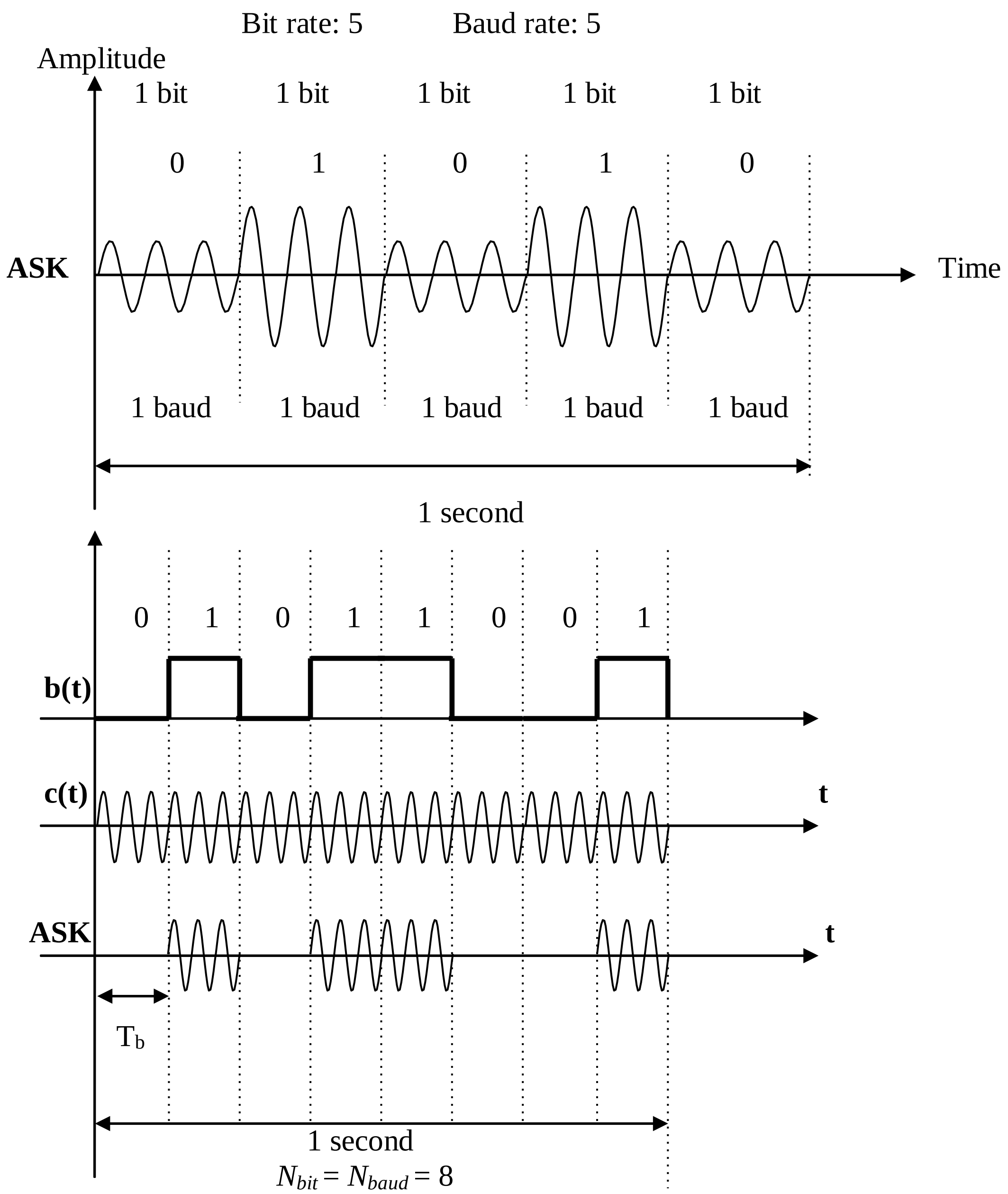
Carrier signal

In analog transmission the sending device produces high - frequency signal that acts as a basis for the information signal. The base signal is called the carrier signal or carrier frequency. The receiving device is tuned to the frequency of the carrier signal that it expects from the sender. Digital information is then encoded onto the carrier signal by modifying one or more of its

characteristic (amplitude, frequency or phase). This kind of modification is called modulation (or shift keying) and the information signal is called a modulating signal.

Amplitude Shift Keying (ASK)

In ASK the strength of the signal is varied to represent binary 1 or 0. Both frequency and phase remain constant, while the amplitude changes.



Bit duration is the period of time that defines one bit. The peak amplitude of the signal, during each bit duration, is constant and its value depends on the bit (0 or 1). The transmission speed using ASK is limited by the physical characteristics of the transmission medium.

andwidth for ASK

$$BW = (1 + d) * N_{baud}$$

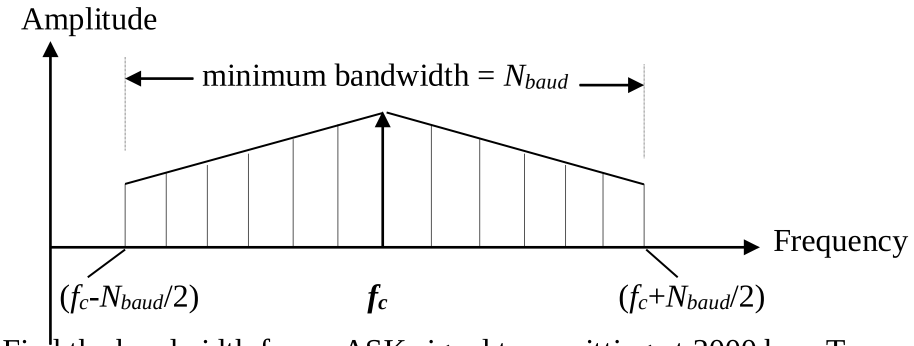
Where

BW is the bandwidth

N_{baud} is the baud rate

d is a factor related to the condition of the line (with min. value of 0)

- The minimum bandwidth required for transmission is equal to the baud rate.



Ex Find the bandwidth for an ASK signal transmitting at 2000 bps. Transmission is in half-duplex mode.

Solution

In ASK baud rate = bit rate

N_{baud} = 2,000

An ASK signal requires a bandwidth equal to its baud rate:

$\Longrightarrow$ BW = 2,000 Hz.

Ex Given a bandwidth of 10,000 Hz (1,000 to 11,000 Hz), draw the full-duplex ASK diagram of the system. Find the carriers and the bandwidth in each direction. Assume there is no gap between the bands in two directions.

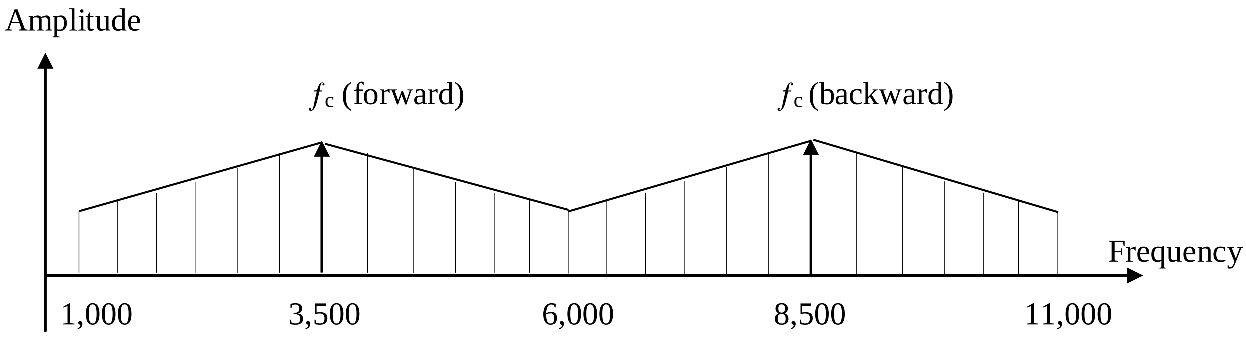
Solution

For full-duplex ASK the bandwidth for each direction is BW=10,000/2=5000 Hz.

The carrier frequencies can be chosen at the middle of each band

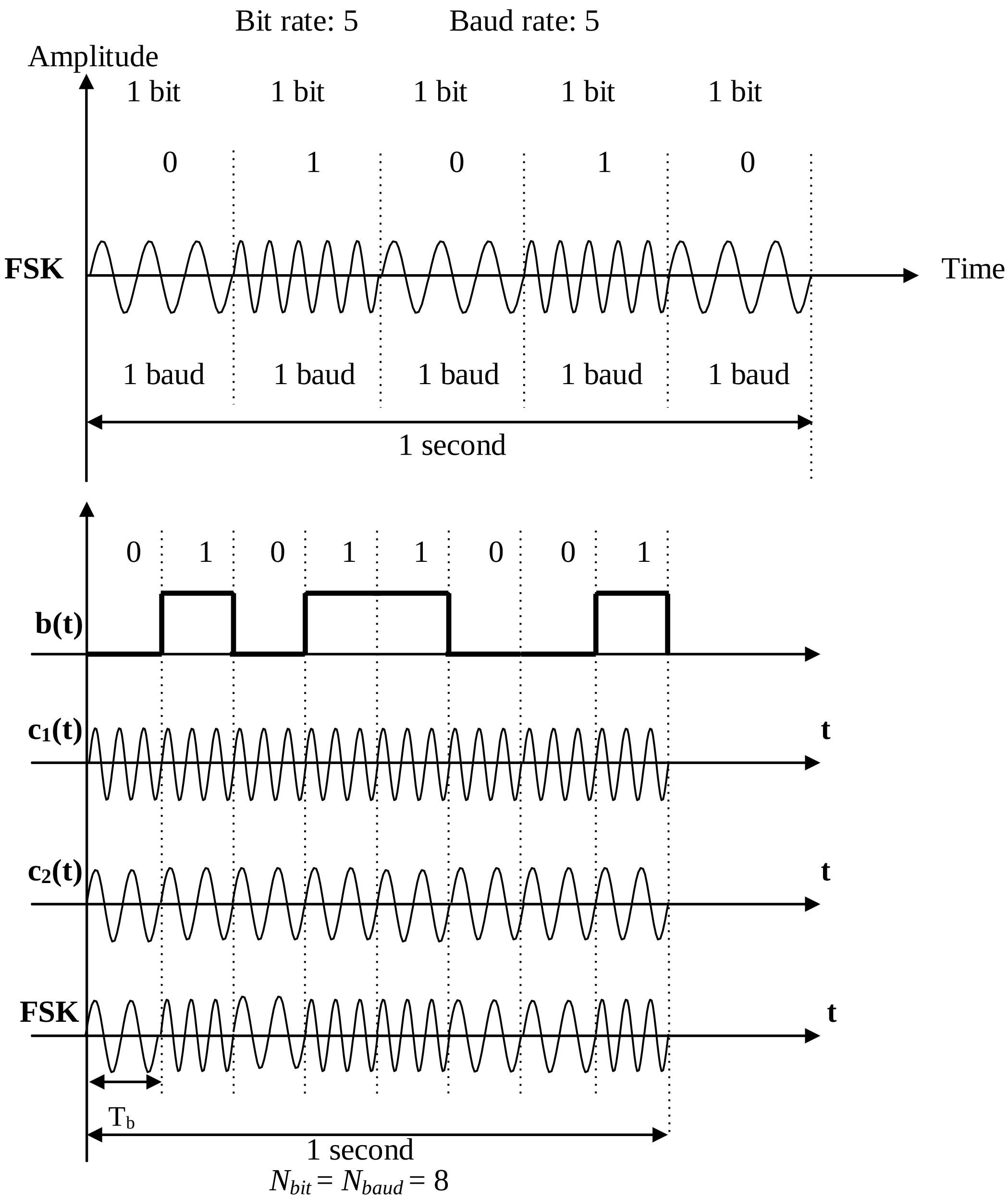
$f_{c\ (forward)}$ = 1,000 + 5,000/2 = 3,500 Hz

$f_{c\ (backward)}$ = 11,000-5,000/2 = 8,500 Hz



Frequency Shift Keying (FSK)

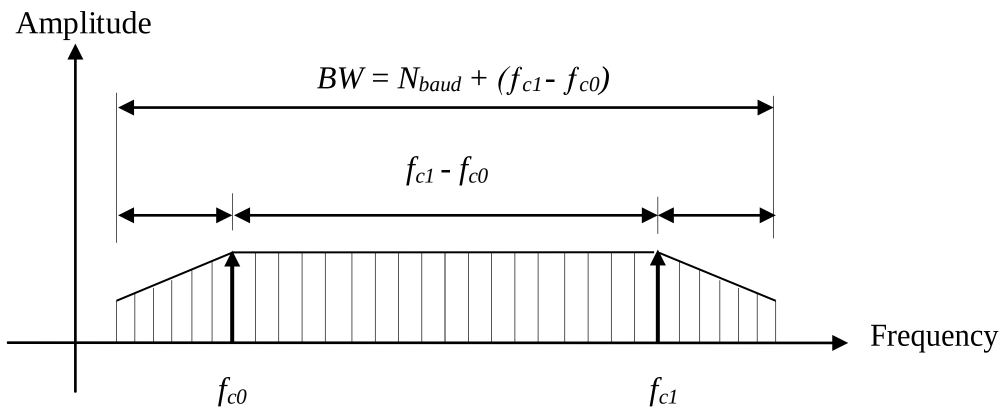
In frequency shift keying (FSK), the frequency of the signal is varied to represent binary 1 or 0. The frequency of the signal during each bit duration is constant and its value depends on the bit (0 or 1): both peak amplitude and phase remain constant.



FSK avoids most of the noise problems of ASK. The limiting factors of FSK are the physical capabilities of the carrier.

Bandwidth for FSK

FSK spectrum can be considered as the combinations of two ASK spectra centred on f_{c0} and f_{c1}. The bandwidth required for FSK transmission is equal to the baud rate of the signal plus the frequency shift (difference between the two carrier frequencies).



$$\boxed{BW = N_{baud} + (f_{c1} - f_{c0})}$$

Ex Find the bandwidth for an FSK signal transmitting at 2,000 bps. Transmission is in half-duplex mode and the carriers must be separated by 3,000 Hz.

Solution

$$BW = N_{baud} + (f_{c1} - f_{c0})$$
$$= 2{,}000 + 3{,}000 = 5{,}000 \text{ Hz.}$$

Ex Find the maximum bit rate for an FSK signal if the bandwidth of the medium is 12,000 Hz and the distance between the two carriers must be at least 2,000 Hz. Transmission is in full-duplex mode.

Solution

Because the transmission is in full-duplex, only 6,000 Hz is allocated for each direction, for FSK, if f_{c1} and f_{c0} are the carrier frequencies,

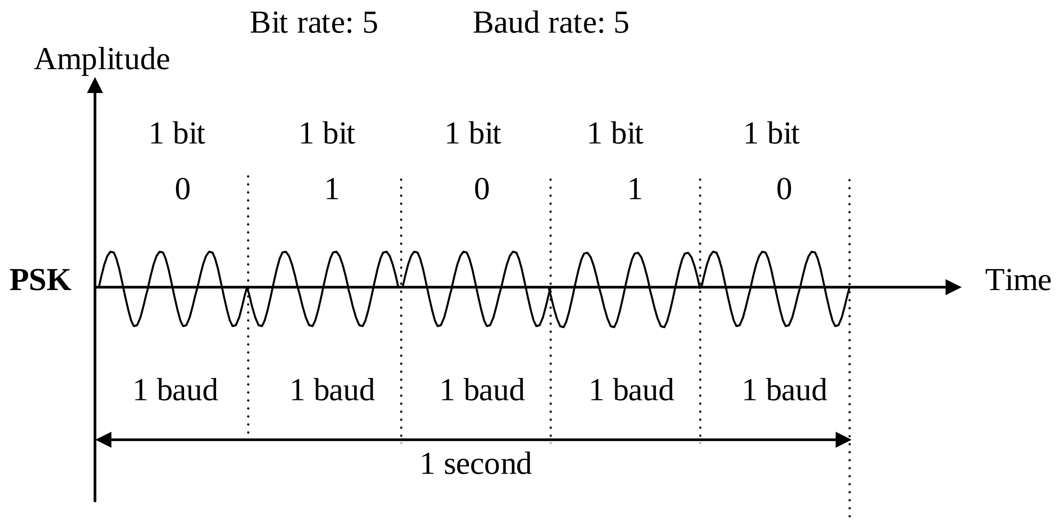
$$BW = N_{baud} + (f_{c1} - f_{c0})$$
$$N_{baud} = BW - (f_{c1} - f_{c0})$$

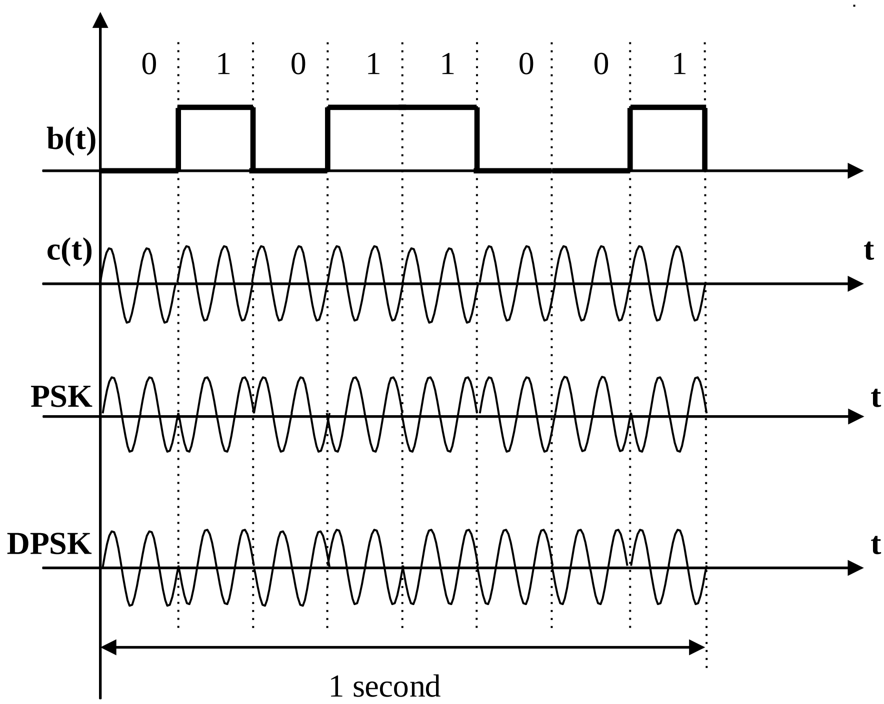
= 6,000 – 2,000 = 4,000

But because the baud rate is the same is bit rate, the bit rate is 4,000 bps.

Phase Shift Keying (PSK)

In the PSK, the phase is varied to represent binary 1 or 0. Both peak amplitude and frequency remain constant as the phase changes. The phase of the signal during each bit duration, is constant and its value depends on the bit (0 or 1).





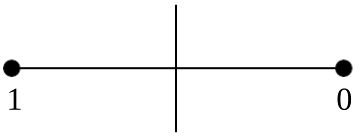
$N_{bit} = N_{baud} = 8$

DPSK eliminates the need for a coherent reference signal at the receiver by combing two basic operations at the transmitter:

- Differential encoding of the input data
- PSK ⟹ DPSK
- To send symbol 1 we phase advance the current signal waveform by 180°.
- To send symbol 0 we leave the phase of the current signal waveform unchanged.

PSK Constellation

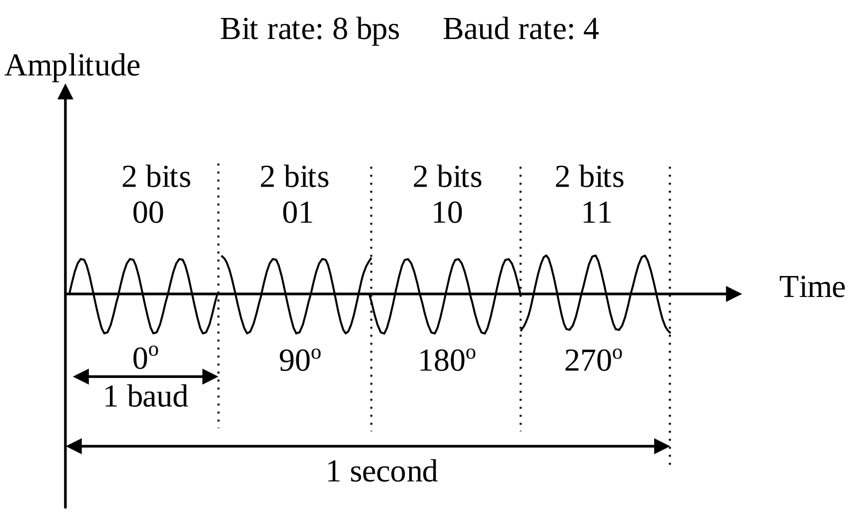
Bit	Phase
0	0^o
1	180°



Constellation diagram

The above method is often called 2-PSK, or binary PSK, because two different phases (0° and 180°) are used in the encoding.

PSK is not susceptible (easily influenced) to the noise degradation that affects ASK, nor to the bandwidth limitations of FSK. This means that smaller variations in the signal can be detected reliably by the receiver. Therefore instead of utilising only two variations of a signal, each representing one bit, we can use four variations and let each phase shift represent two bits.



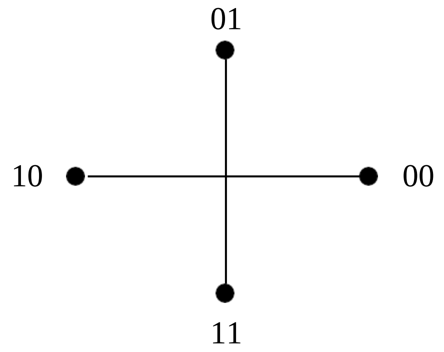
4-PSK (Quadrature-PSK)

This technique is called 4-PSK or Q-PSK. The pair of bits represented by each phase is called a dibit.

Data can be transmitted two times as fast using 4-PSK as using 2-PSK.

4–PSK characteristics

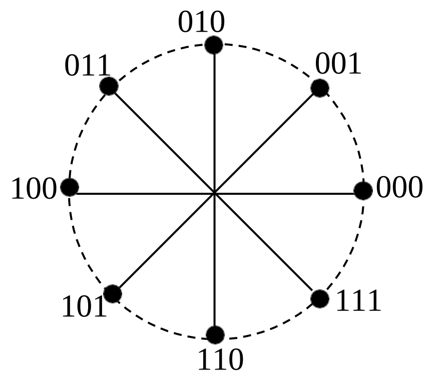
Dibit	Phase
00	0^o
01	90^o
10	180^o
11	270^o



Constellation diagram

8–PSK Characteristics

Tribit	Phase
000	0^o
001	45^o
010	90^o
011	135^o
100	180^o
101	225^o
110	270^o
111	315^o



Constellation diagram

Bit rate of 8-PSk is three as that of 2-PSK

Bandwidth for PSK

The min. BW required for PSK transmission is the same as that required for ASK transmission.

$$BW = N_{baud}$$

Ex: Find the bandwidth for a 4-PSK signal transmitting at 2,000 bps. Transmission is in half-duplex mode.

Solution

For 4-PSK the baud rate is half of the bit rate.

The baud rate is therefore 1,000. A PSK signal requires a bandwidth equal to its baud rate.

Therefore the bandwidth is 1,000 Hz.

Ex: Given a bandwidth of 5,000 Hz for an 8-PSK signal, what are the baud rate and bit rate?

Solution

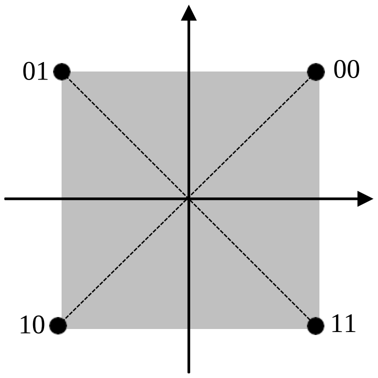
For PSK the baud rate is the same as the bandwidth, which means the baud rate is 5,000. But in 8-PSK, the bit rate is three times the baud rate. So the bit rate is 15,000 bps.

Quadrature Amplitude Modulation (QAM)

QAM means combining ASK and PSK in such a way that we have maximum contrast between each bit, dibit, tribit, quadbit, and so on.

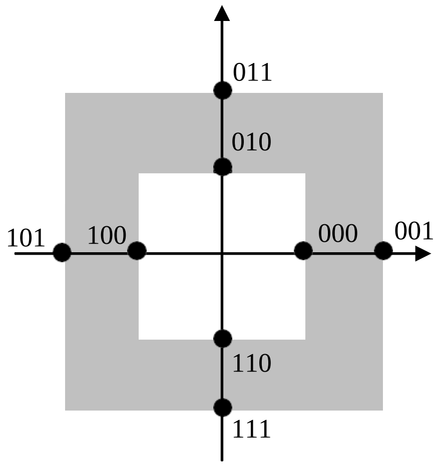
Possible variation of QAM is numerous; theoretically any measurable number of changes in amplitude can be combined with any measurable number of changes in phase.

The figure below shows the constellation diagrams of 4-QAM and 8-QAM. In both cases the number of amplitude shifts is less than the number of phase shifts. Because amplitude changes are susceptible to noise and require greater shift differences than do phase changes, the number of phase shifts used by a QAM system is always larger than the number of amplitude shifts.



4-QAM

Constellation Diagram

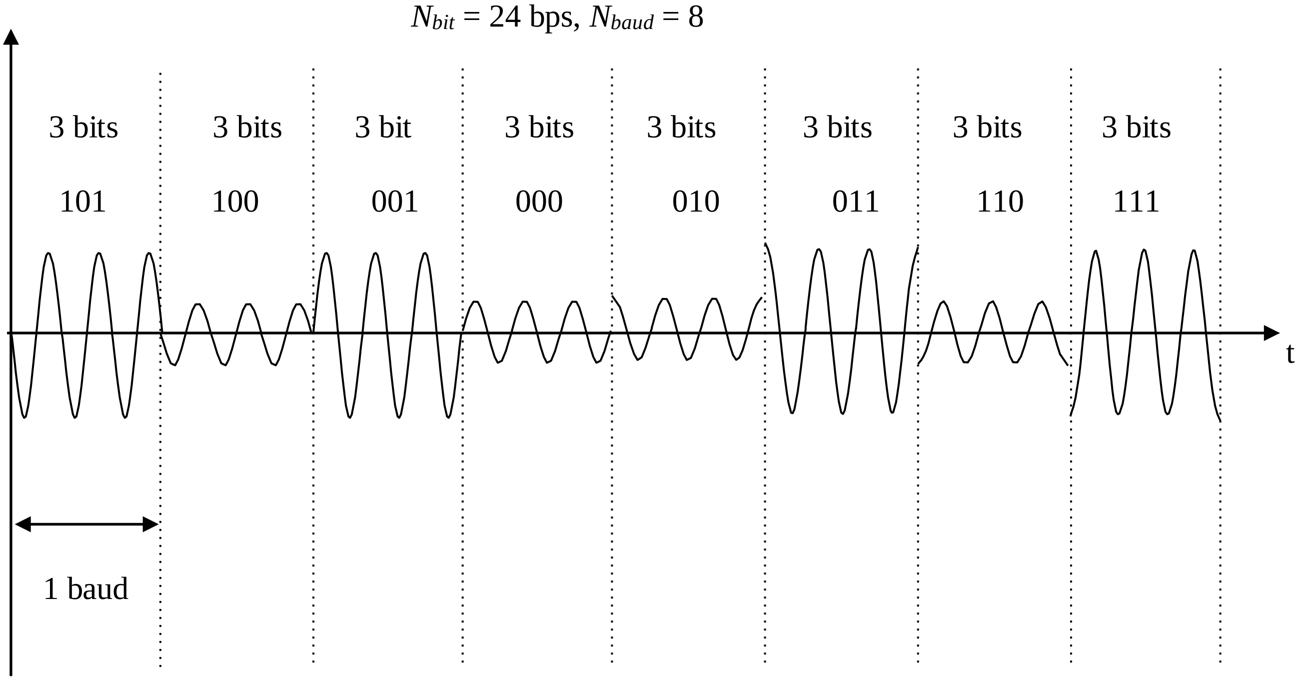


8-QAM

Constellation Diagram

Amplitude

N_{bit} = 24 bps, N_{baud} = 8

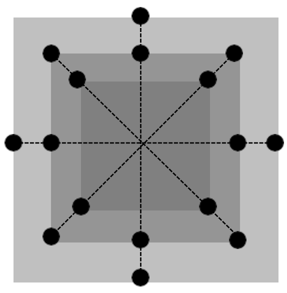


The output signal of the 8-QAM modem for data

b (t) = 101 100 001 000 010 011 110 111

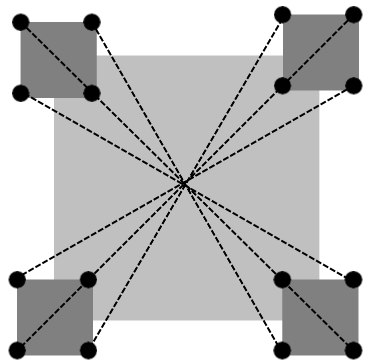
Three popular 16-QAM configurations are shown bellow:

4 amplitudes, 8 phase



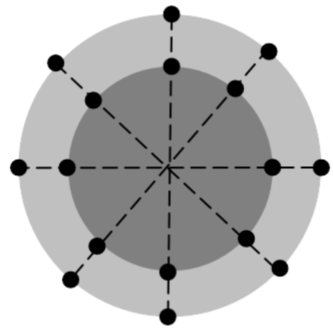
16-QAM

3 amplitudes, 12 phases



16-QAM

2 amplitude, 8 phase



16-QAM

Since amplitude shift is more susceptible to noise, the greater the ratio of phase shifts to amplitude, the greater the immunity to noise.

The second example, three amplitudes and 12 phases, handles noise best. The first example, 4 amplitudes and 8 phases, is the OSI (Open Systems Interconnection) recommendation. Several QAM designs link specific amplitudes with specific phases. This means that even with noise problems associated with amplitude shifting, the meaning of a shift can be recovered from phase information.

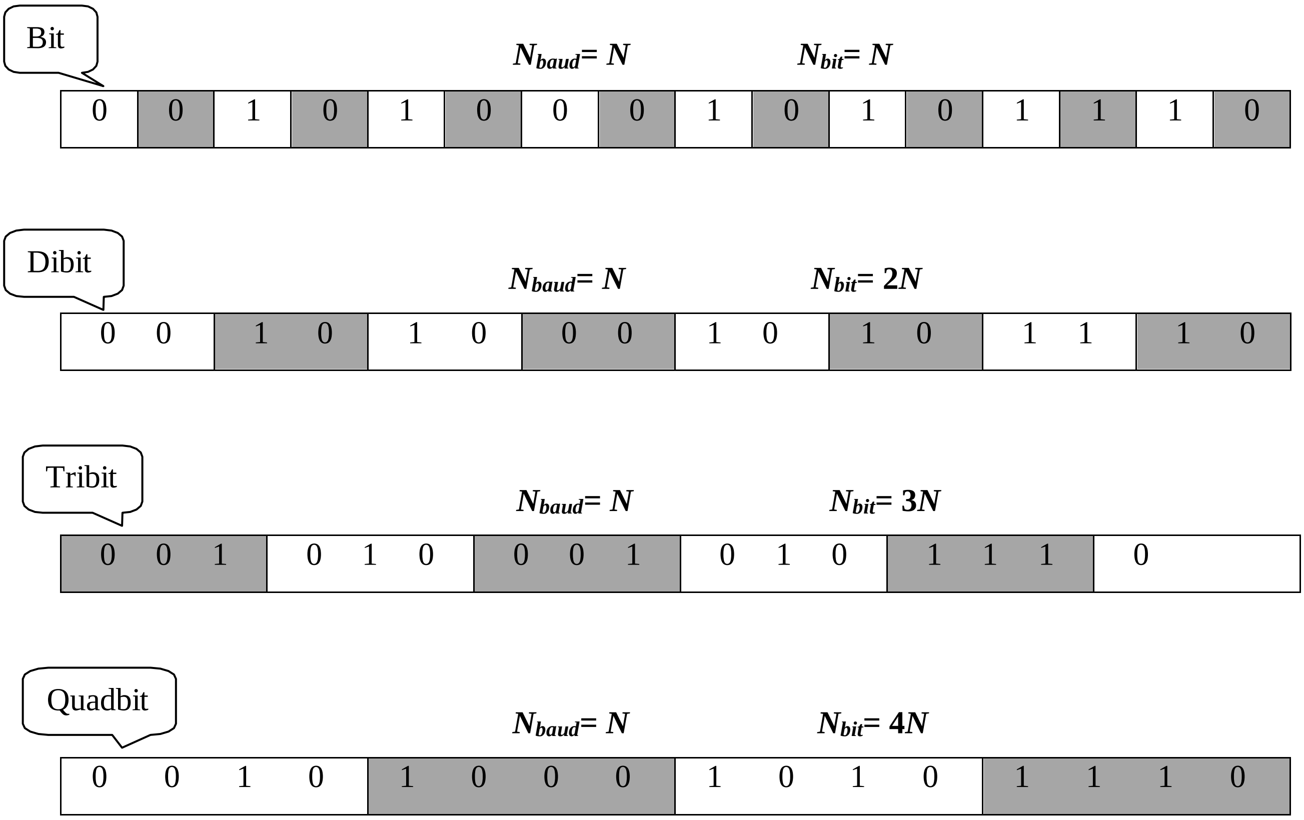
Bandwidth for QAM

The minimum bandwidth required for QAM transmission is the same as that required for ASK and PSK transmission. QAM has the same advantages of PSK over ASK.

Bit/Baud Comparison

Assuming that an FSK signal over voice-grade phone lines can send 1200 bps, the bit rate is 1200 bps. Each frequency shift represents a single bit; so it requires 1200 signal elements to send 1200 bits. Its baud rate, therefore, is also 1200. Each signal variation in 8-QAM system, however, represents three bits. So a bit rate of 1200 bps, using 8-QAM, has a baud rate of only 400.

As the figure below shows, a dibit system has a baud rate of one-half the bit rate, a tribit system has a baud rate of one-third the bit rate, a quadbit system has a baud rate of one-fourth the bit rate.



Bit/ Baud Rate Comparison

Modulation	*Units*	*Bits/Bauds*	*Baud Rate*	*Bit Rate*
ASK, FSK, 2-PSK	**Bit**	**1**	**N**	**N**
4-PSK, 4-QAM	**Dibit**	**2**	**N**	**2N**
8-PSK, 8-QAM	**Tribit**	**3**	**N**	**3N**
16-QAM	**Quadbit**	**4**	**N**	**4N**
32-QAM	**Pentabit**	**5**	**N**	**5N**
64-QAM	**Hexabit**	**6**	**N**	**6N**
128-QAM	**Septabit**	**7**	**N**	**7N**
256-QAM	**Octabit**	**8**	**N**	**8N**

Ex

A constellation diagram consists of eight equally spaced points on a circle. If the bit rate is 4800 bps, what is the baud rate?

Solution

The constellation indicates 8-PSK with points 45° apart. Since $2^3 = 8$, three bits are transmitted with each signal element. Therefore, the baud rate is

$$4800/3 = 1600 \text{ baud}$$

Ex

Compute the baud rate for a 72,000 bps 64-QAM.

Solution

A 64-QAM signal means that there are six bits per signal elements since $2^6 = 64$. Thus,

$$72{,}000/6 = 12{,}000 \text{ baud}$$

Ex

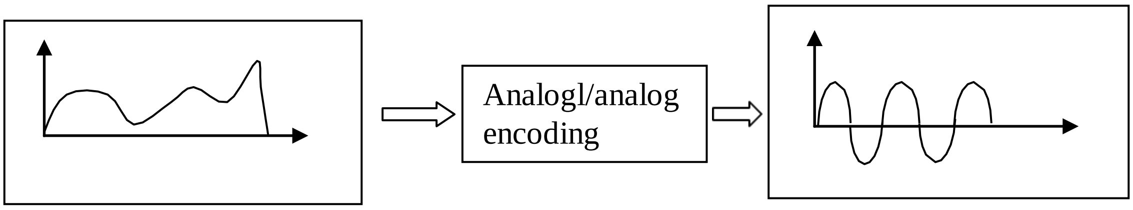
Compute the bit rate for a 1,000 baud 16-QAM signal.

Solution

A 16-QAM signal means that there are four bits per signal elements since $2^4 = 16$. Thus,

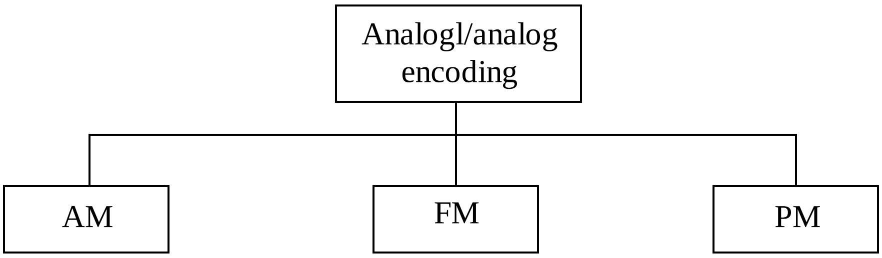
$$(1{,}000)(4) = 4{,}000 \text{ bps}$$

Analog-to-Analog-Encoding



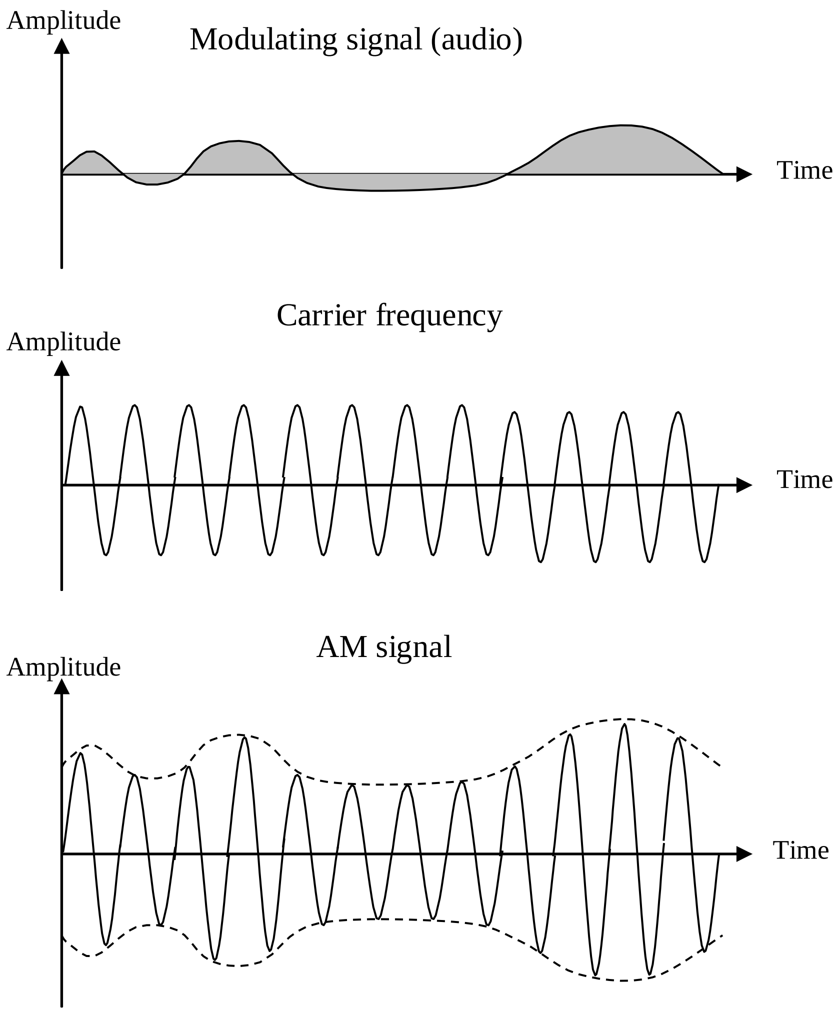
Analog-to-analog encoding is the representation of analog information by an analog signal. (eg. Radio communication).

Analog-to-analog modulation can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM)



Amplitude Modulation

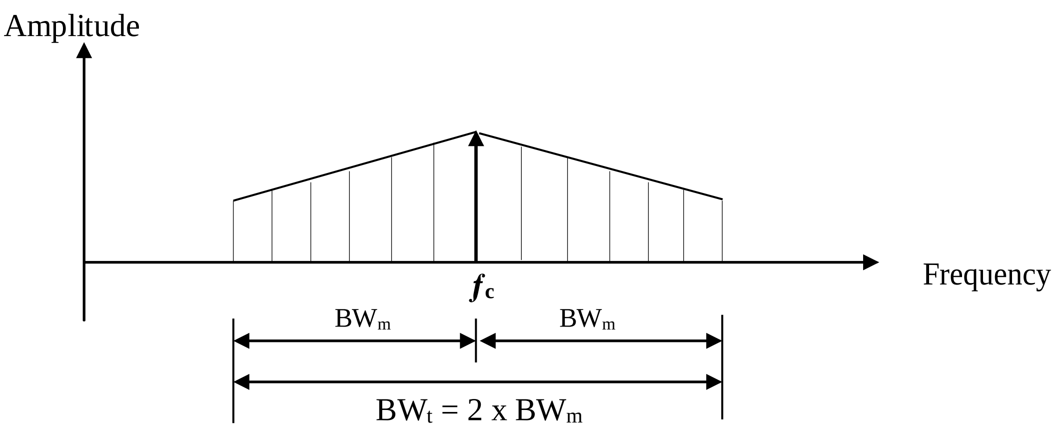
In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information. The modulating signal becomes the envelope of the carrier.



AM Bandwidth

The bandwidth of an AM signal is equal to twice the bandwidth of the modulating signal and covers a range centred on the carrier frequency. The total bandwidth required for AM can be determined from the bandwidth of the audio signal:

$$\mathbf{BW_t = 2 \times BW_m}$$



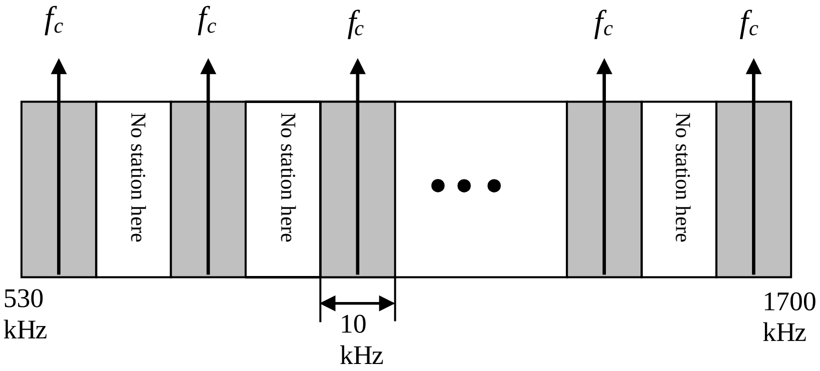
BW_m = Bandwidth of the modulating signal (audio)

BW_t = Total bandwidth (radio)

f_c = Frequency of the carrier

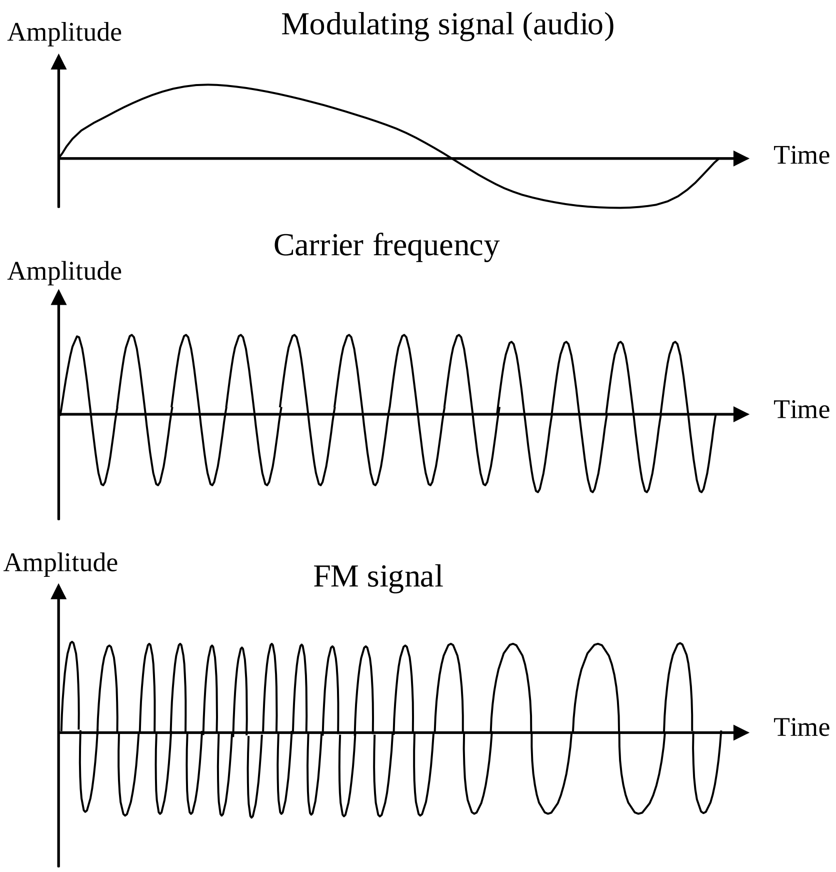
The bandwidth of an audio signal (speech & music) is usually 5 kHz. Therefore, an AM radio station needs a minimum bandwidth of 10 kHz. In fact, the Federal Communications Commission (FCC) allows 10 kHz for each AM station.

AM stations are allowed carrier frequencies anywhere between 530 and 1700 kHz (1.7 MHz). However, each station's carrier frequency must be separated from those on either side by at least 10 kHz (one AM bandwidth) to avoid interference.



Frequency Modulation (FM)

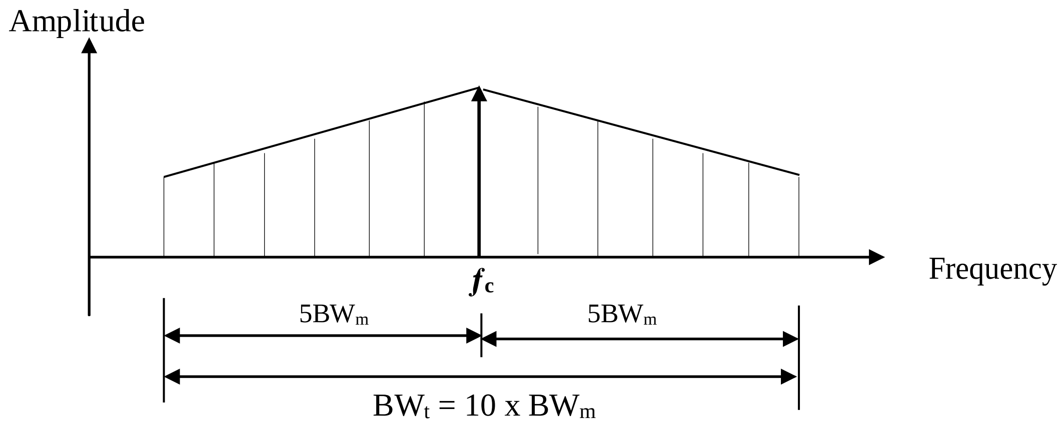
In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.



FM Bandwidth

The bandwidth of an FM signal is equal to 10 times the bandwidth of the modulating signal and, like AM bandwidth, covers a range centred on the carrier frequency. The total bandwidth required for FM can be determined from the bandwidth of the audio signal:

$$\mathbf{BW_t = 10 \times BW_m}$$



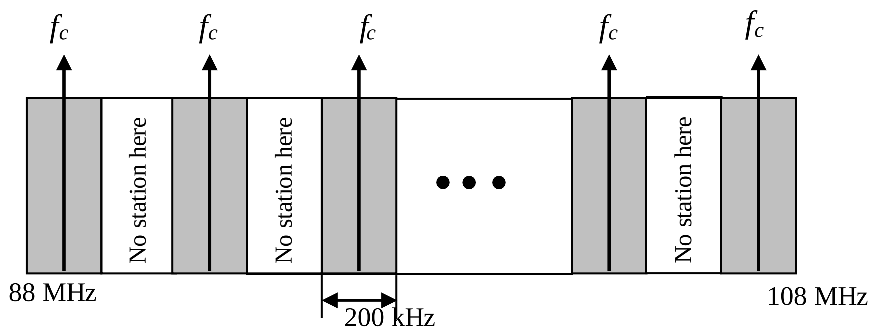
BW_m = Bandwidth of the modulating signal (audio)

BW_t = Total bandwidth (radio)

f_c = Frequency of the carrier

The bandwidth of an audio signal (speech & music) broadcast in stereo is almost 15 kHz. Therefore, each FM radio station needs a minimum bandwidth of 150 kHz. The FCC allows 200 kHz (0.2 MHz) for each FM station to provide some room for guard bands.

FM stations are allowed carrier frequencies anywhere between 88 and 108 MHz. However, stations must be separated from by at least 200 kHz to avoid overlapping.



Phase Modulation (PM)

Due to simpler hardware requirements, PM is used in some systems as an alternative to FM. In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of carrier changes correspondingly. The analysis and final result (modulating signal) are similar to those of FM.

TRANSMISSION CODES

Transmission Codes

Binary-Coded Decimal (also called 8421 BCD)

In BCD, four bits are used to encode one decimal character. Four bits give 16 binary combinations. Since there are 10 decimal characters, 0 through 9, only 10 of the 16 possible combinations are necessary for encoding in BCD. The remaining 6 combinations are said to be invalid.

Decimal	BCD
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001

1010
1011
1100
1101
1110
1111

Not valid in BCD

Ex

Convert 367_{10} to BCD

Solution

$$367_{10} = 0011\ 0110\ \ 0111$$

Ex

Convert 1249_{10} to BCD

Solution

$$1249_{10} = 0001\ 0010\ 0100\ \ 1001$$

Ex

Convert 58_{10} to BCD

Solution

$$58_{10} = 0101\ \ 1000$$

BCD Addition:

Straight binary addition is performed as long as the result does not exceed a decimal value of 9.

Ex

Add the decimal numbers 3 and 4 in BCD

Solution

3	0011
+ 4	0100
7	0111

Ex

Add the decimal numbers 63 and 24 in BCD

Solution

63	0110 0011
+ 24	0010 0100
87	1000 0111

When the sum of two numbers exceeds 9, an invalid BCD number is obtained. The invalid number can be converted to a valid number by adding 0110 (6) to it.

Ex

Add the decimal numbers 9 and 6 in BCD

Solution

9	1001
+ 6	0110
15	1111…not valid in BCD
	+ 0110…add 6 for correction
	0001 0101…correct BCD number

Ex

Add the decimal numbers 46 and 79 in BCD

Solution

46	0100 0110
+ 79	0111 1001
125	1011 1111
	+ 0110 0110
	0001 0010 0101
	1 2 5

Excess-3 Code

Excess-3 code is very similar to 8421 BCD code. The only difference is that 3 is added to the decimal before it is encoded into a four-word.

Decimal	BCD	Excess-3		
0	0000	0011		
1	0001	0100		
2	0010	0101	0000	
3	0011	0110	0001	
4	0100	0111	0010	
5	0101	1000	1101	Not valid in Excess-3
6	0110	1001	1110	
7	0111	1010	1111	
8	1000	1011		
9	1001	1100		

Ex

Add the decimal numbers 9 and 7 in Excess-3

Solution

9	1100
+ 7	1010
16	1 0110

Gray Code

The disadvantage of the previous codes is that several bits change state between adjacent counts. The Gray code is unique in that successive counts result in only one bit change. For example, 7 (0111) to 8 (1000) in binary, or BCD, all four bits change state. In Gray code, however, 7 (0100) to 8 (1100) require a single bit change.

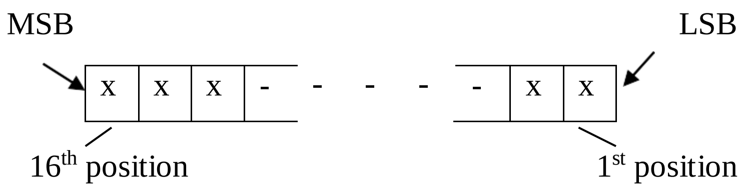
The switching noise generated by the associated circuits may be intolerable in some environments. The same change with Gray code undergoes only a single bit change consequently, less noise is generated. Shaft encoders used for receiver tuning dials often use Gray code.

The Gray code is widely used for encoding the position of the rotary shaft and for data transmission using PSK.

Decimal	Binary	Gray Code
0	0000	0000
1	0001	0001
2	0010	0011
3	0011	0010
4	0100	0110
5	0101	0111
6	0110	0101
7	0111	0100
8	1000	1100
9	1001	1101
10	1010	1111

Binary-to-Gray Conversion

- The given binary code is shifted to the right by one bit.
- Discard the last bit (the LSB) from the obtained bits.
- Exclusive-ORing the given and obtained bits result in the equivalent Gray code.



Ex

Compute the Gray code for the binary number 11010

Solution

Binary code		1 1 0 1 0
	$\oplus$	1 1 0 1
Gray code		**1 0 1 1 1**

Ex

Compute the Gray code for the binary number 10001101

Solution

Binary code		1 0 0 0 1 1 0 1
	$\oplus$	1 0 0 0 1 1 0
Gray code		**1 1 0 0 1 0 1 1**

Gray-to-Binary Conversion

- The first bit, the leftmost of the given Gray code, becomes the MSB of the Binary code.
- Exclusive-ORing the second Gray code bit with the MSB of the binary code yields the second binary bit.
- Exclusive-ORing the third Gray code bit with the second binary code yields the third binary bit.
- Exclusive-ORing the fourth Gray code bit with the third binary code yields the fourth binary bit. And so on.

Ex

Compute the binary code for the Gray code 101101

Solution

Gray code	1	0	1	1	0	1
Binary code	**1**	**1**	**0**	**1**	**1**	**0**

Ex

Compute the binary code for the Gray code 11001011

Solution

Gray code	1	1	0	0	1	0	1	1
Binary code	**1**	**0**	**0**	**0**	**1**	**1**	**0**	**1**

Binary Numbers

The binary numbering system provides the basis for all computer operations. Computers work by manipulating electrical current on and off. The binary system uses two symbols, 0 and 1. Also called base 2.

Binary weights

Position	**Fifth**	**Fourth**	**Third**	**Second**	**First**
Weight	2^4 (16)	2^3 (8)	2^2 (4)	2^1 (2)	2^0 (1)

1	1	0	1	digits
8	4	2	1	weights
8	4	0	1	results

+

13

Octal Numbers

The octal numbering system is used by computer programmers to represent binary numbers in compact form. Also called base 8.

Octal numbers use 8 symbols: 0,1,2,3,4,5,6,7.

Octal weights

Position	**Fifth**	**Fourth**	**Third**	**Second**	**First**
Weight	8^4 (4096)	8^3 (512)	8^2 (64)	8^1 (8)	8^0 (1)

Ex

3	4	7	1	digits
512	64	8	1	weights
1,536	256	56	1	results

+

1,849

Hexadecimal Numbers

Hexadecimal numbering system, like octal, is used by computer programmers to represent binary numbers in compact form. Also called base 16.

Hexadecimal uses 16 symbols: 0,1,2,3,4,5,6,7,8,9,A,B,C,D,E,F.

Hexadecimal weights

Position	**Fifth**	**Fourth**	**Third**	**Second**	**First**
Weight	16^4 (65,536)	16^3 (4,096)	16^2 (256)	16^1 (16)	16^0 (1)

3	4	7	1	digits
4,096	256	16	1	weights
12,288	1,024	112	1	results

+

13,425

Decimal	Binary	Octal	Hexadecimal
0	0000	0	0
1	0001	1	1
2	0010	2	2
3	0011	3	3
4	0100	4	4
5	0101	5	5
6	0110	6	6
7	0111	7	7
8	1000	10	8
9	1001	11	9
10	1010	12	A
11	1011	13	B
12	1100	14	C
13	1101	15	D
14	1110	16	E
15	1111	17	F

<u>Transformations</u>

- From Other Systems to Decimal

a) From Binary to Decimal

1	0	0	1	1	1	0	Binary
64	32	16	8	4	2	1	weights
64	0	0	8	4	2	0	weighted results

+

78

b) Hexadecimal to Decimal

4	E	Hexadecimal
16	1	weights
64	14	weighted results

+

78

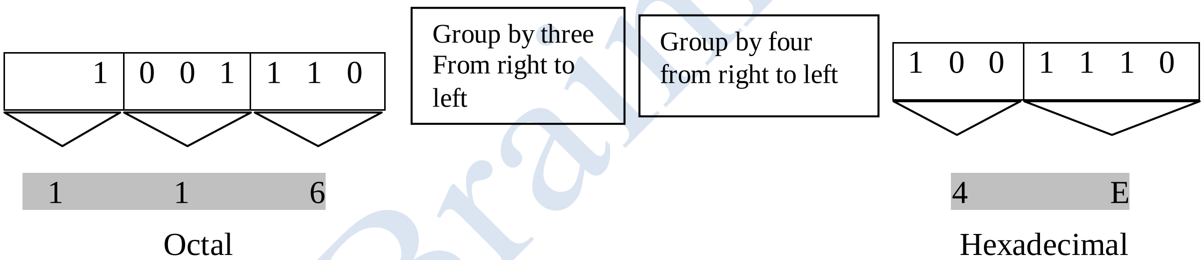
c) Octal to Decimal

1	1	6	Octal
64	8	1	weights
64	8	6	weighted results

+

78

- From Binary to Octal or Hexadecimal

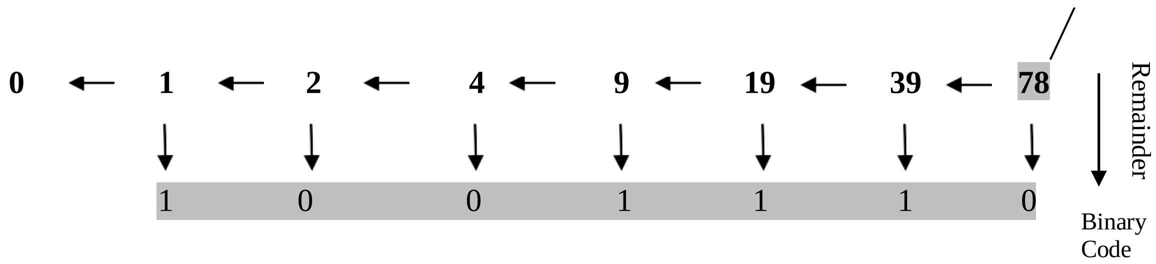


- From Octal or Hexadecimal to Binary

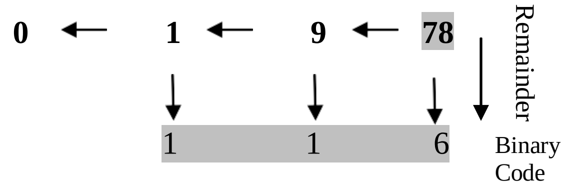
- From Decimal to Other Systems

a) From Decimal to Binary (divide by 2)

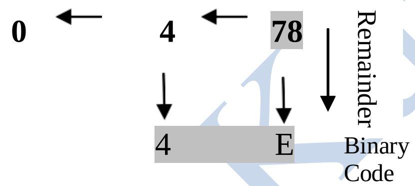
Decimal



b) From Decimal to Octal (divide by 8)



c) From Decimal to Hexadecimal (divide by 16)



Morse Code

Morse code is one of the oldest electrical transmission codes. The digital code system is made up of a series of dots and dashes, representing the alphabet and decimal numbers system. A dash is three times the duration of a dot.

A ∘ —
B — ∘ ∘ ∘
C — ∘ — ∘
D — ∘ ∘
.

1 ∘ — — — — —
2 ∘ ∘ — — —
.

ASCII Code

The American Standard Code for Information Interchange (ASCII) is the most widely used alphanumeric code for transmission and data processing.

ASCII is a seven-bit code that can be represented by two hexadecimal characters for simplicity. The MS hexadecimal character in this case never exceed 7

Ex

What is the ASCII code for the letter H (uppercase) in binary and hexadecimal?

Solution

Letter H is located in column 4 and row 8.

Binary: 100 1000

Hexadecimal: $ 48

Ex

What is the ASCII code for the letter k (lowercase) in binary and hexadecimal?

Solution

Letter k is located in column 6 and row B:

Binary: 110 1011

Hexadecimal: $ 6B